



Application Notes for Configuring Avaya Aura® Solution for Midsize Enterprise 6.2.2 and Avaya Session Border Controller for Enterprise 6.2 with Alestra Enlace IP SIP Trunk Service – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between the Alestra Enlace IP SIP Trunk Service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Solution for Midsize Enterprise 6.2.2 and Avaya Session Border Controller for Enterprise 6.2.

The Enlace IP SIP trunk service offered by Alestra provides customers with PSTN access via a SIP trunk between the enterprise and the Alestra network, as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between the Alestra Enlace IP SIP trunk service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of a HP® Proliant DL360 server running the Avaya Aura® Solution for Midsize Enterprise 6.2.2 template, an Avaya Session Border Controller for Enterprise (Avaya SBCE) 6.2 and various Avaya endpoints.

The Alestra Enlace IP SIP Trunk Service referenced within these Application Notes is designed for enterprise business customers in Mexico. Customers using this service with the Avaya SIP-enabled enterprise solution are able to place and receive PSTN calls via a broadband WAN connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as analog and/or ISDN-PRI.

2. General Test Approach and Test Results

A simulated enterprise site containing all the equipment for the Avaya SIP-enabled solution was installed at the Avaya Solution and Interoperability Lab. The enterprise site was configured to connect to the Alestra Enlace IP SIP Trunk Service via a broadband connection to the public Internet.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

To verify SIP trunking interoperability, the following features and functionality were covered during the interoperability compliance test:

- Response to SIP OPTIONS queries.
- Incoming PSTN calls to various phone types. Phone types included H.323, SIP, digital, and analog telephones at the enterprise. All inbound calls from the PSTN were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls from various phone types. Phone types included H.323, SIP, digital, and analog telephones at the enterprise. All outbound calls to the PSTN were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from Avaya one-X® Communicator softphones.
- Avaya one-X® Communicator supports placing and receiving calls using the local computer or by controlling an external telephone. Usage modes “This Computer” and “Other Phone” were tested. Avaya one-X® Communicator also supports two signaling protocols: H.323 and SIP. Each supported protocol was tested.
- Inbound and outbound PSTN calls to/from Avaya Flare® Experience for Windows softphones.
- Various call types, including: local, long distance and international.
- Codecs G729A and G.711A and proper codec negotiation.
- DTMF tone transmissions passed as out-of-band RTP events as per RFC 2833.
- Caller ID presentation and Caller ID restriction.
- Voicemail redirection and navigation.
- User features such as hold and resume, transfer, and conference.
- Off-net call forwarding and mobility (extension to cellular).
- Routing inbound PSTN calls to call center agent queues.
- T.38 fax.

Items not supported or not tested included the following:

- Inbound toll-free and emergency calls are supported but were not tested as part of the compliance test
- Network Call Redirection methods using REFER or 302 Temporarily Unavailable messages are not supported by Alestra and were not tested.
- Operator services such as dialing 0 or 0 + 10 digits are not supported.

2.2. Test Results

Interoperability testing of the Alestra Enlace IP SIP trunk service with the Avaya SIP-enabled enterprise solution was completed with successful results for all test cases with the exception of the observations and limitations described below:

- **Calls to Busy Numbers:** Alestra does not send “486 Busy Here” for calls from an enterprise extension to busy PSTN numbers. Since busy tone is heard by the caller this observation has no impact to the user.
- **Caller ID on incoming calls from the U.S.:** Calls originating from PSTN telephones in the U.S. to DID numbers in Mexico assigned to the SIP trunk to the Avaya solution will display **Restricted/Unavailable** on the enterprise extensions. This seems to be a PSTN restriction for all calls from the U.S. to Mexico, not limited just to Alestra. This behavior is not necessarily indicative of a limitation of the combined Alestra/Avaya solution, and it is listed here simply as an observation.

2.3. Support

For technical support on the Alestra Enlace IP SIP trunk service offer, visit <http://www.alestra.com.mx/>

3. Reference Configuration

Figure 1 illustrates the sample Avaya SIP-enabled enterprise solution, connected to the Alestra Enlace IP SIP trunk service through a public Internet WAN connection.

For security purposes, private addresses are shown in these Application Notes for the Avaya SBCE and the Service Providers public network interfaces, instead of the real public IP addresses used during the tests. Also, PSTN routable phone numbers used in the compliance test have been changed to non-routable numbers.

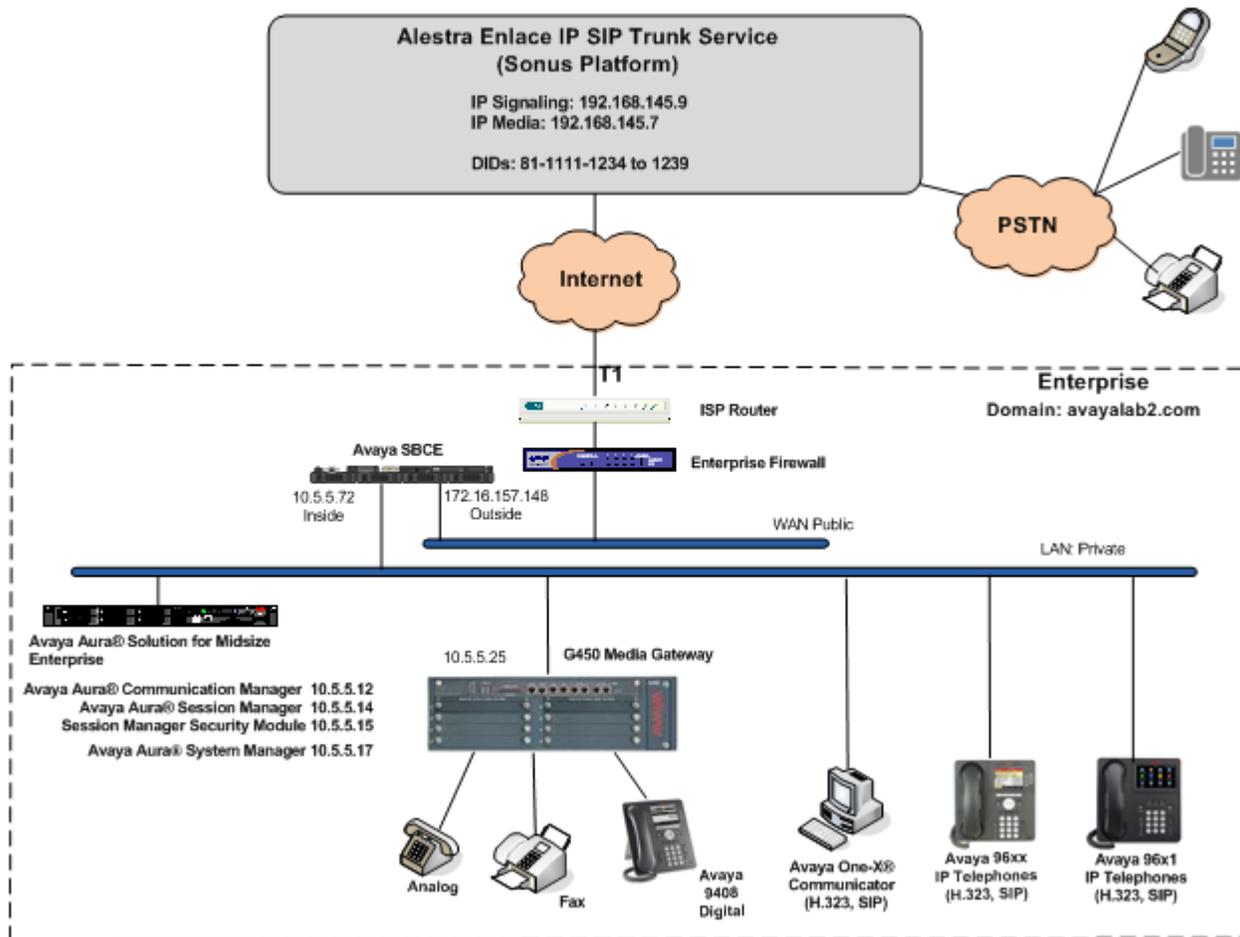


Figure 1: Avaya SIP Enterprise Solution connected to Alestra Enlace IP SIP Trunk Service

The Avaya components used to create the simulated customer site included:

- Avaya Aura® Solution for Midsize Enterprise 6.2.2
- Avaya Session Border Controller for Enterprise 6.2
- Avaya G450 Media Gateway
- Avaya 96x0 and 96x1 Series IP Telephones (H.323 and SIP)
- Avaya one-X® Communicator soft phones (H.323 and SIP)

- Avaya Flare® Experience for Windows softphones (SIP)
- Avaya digital and analog telephones

The Avaya Aura® Solution for Midsize Enterprise Release 6.2.2 Template delivers the following applications as virtual machines running on top of System Platform 6.3.0:

- Communication Manager 6.3.0
- Communication Manager Messaging 6.3.0
- Session Manager 6.3.2
- System Manager 6.3.2
- Presence Services 6.2.0
- Utility Services 6.3.0
- Application Enablement Services 6.3.0

These Application Notes will not cover the software installation of System Platform and the loading of the Midsize Enterprise Template. Consult the documentation on the **References** section for more information and instructions on the software installation of System Platform, the Midsize Enterprise Template and initial configuration.

The following screen shows the list of the applications installed and running on the server, as seen from the Virtual Machine Management screen in System Platform.

Avaya Aura® System Platform
admin
Previous successful login: Wed Aug 14 17:43:16 EDT 2013
Failed login attempts since: 1
HA status: **Not configured**
About | Help | Log Out

Virtual Machine Management
Virtual Machine List ⓘ
System Domain Uptime: 14 days, 3 hours, 44 minutes, 47 seconds
Current template installed: Midsize_Ent 6.2.2.0.1120 (smgr 6.3.8.0, aes 6.3.0.0.212, cm 06.3-03.0.124.0, utility_server 6.3.0.0.20, sm 6.3.2.0.632028, presence_va 6.2.0.0.0182) Refresh

Name	Version	IP Address	Max Memory	Virtual CPUs	CPU Time	State	Application State
Domain-0	6.3.0.0.18002	10.5.5.10	864.0 MB	12	1d 16h 3m 39s	Running	N/A
sm	6.3.2.0.632028	10.5.5.14	6.0 GB	6	1d 0h 26m 51s	Running	Running
cm	06.3-03.0.124.0	10.5.5.12	4.5 GB	1	12h 49m 42s	Running	Running
aes	6.3.0.0.212	10.5.5.18	4.0 GB	4	12h 26m 24s	Running	Running
cdom	6.3.0.0.18002	10.5.5.11	512.0 MB	1	15h 6m 59s	Running	N/A
utility_server	6.3.0.0.20	10.5.5.13	1024.0 MB	1	2h 51m 51s	Running	Running
presence_va	6.2.0.0.0182	10.5.5.16	12.0 GB	6	4h 20m 53s	Running	N/A
smgr	6.3.2.4.1529	10.5.5.17	9.0 GB	4	1d 22h 7m 16s	Running	Running

Note: Application Enablement and Presence Services are installed as part of the Midsize Enterprise Template, but since these applications were not used during the compliance testing, the configuration of these services is not covered in these Application Notes.

Located at the edge of the enterprise is the Avaya SBCE. It has a public side that connects to the external network and a private side that connects to the enterprise infrastructure. All SIP and RTP traffic entering or leaving the enterprise flows through the Avaya SBCE. In this way, the Avaya SBCE can protect the enterprise against any SIP-based attacks. The Avaya SBCE also performs network address translation at both the IP and SIP layers.

The transport protocol between the Avaya SBCE and Alestra across the public IP network is UDP. The transport protocol between the Avaya SBCE and the enterprise Session Manager across the enterprise IP network is TCP.

For inbound calls, the calls flow from the service provider to the external firewall, to the Avaya SBCE, then to Session Manager. Session Manager uses the configured dial patterns (or regular expressions) and routing policies to determine the recipient (in this case the Communication Manager) and on which link to send the call. Once the call arrives at Communication Manager, further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed.

Outbound calls to the PSTN were first processed by Communication Manager for outbound feature treatment such as automatic route selection and class of service restrictions. Once Communication Manager selected the proper SIP trunk, the call was routed to Session Manager. Session Manager once again used the configured dial patterns (or regular expressions) and routing policies to determine the route to the Avaya SBCE for egress to the Alestra network.

A separate SIP trunk was created between Communication Manager and Session Manager to carry the service provider traffic. This was done so that any trunk or codec settings required by the service provider could be applied only to this trunk without affecting other enterprise SIP traffic. The trunk carried both inbound and outbound traffic.

Communication Manager Messaging, defined as a separate SIP entity in Session Manager, was used during the compliance test to verify voice mail redirection and navigation, as well as the delivery of MWI (Message Waiting Indicator) messages to the enterprise telephones. Since the configuration tasks for Communication Manager Messaging are not directly related to the interoperability tests with the Alestra Enlace IP SIP trunk service, they are not included in these Application Notes.

During the compliance test, in addition to the DID numbers assigned to the SIP trunk, Alestra provided a local test number in Monterrey, Mexico. A SIP-based softphone was registered to this local PSTN number and was used to originate and terminate local calls to and from the PSTN to the enterprise.

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Component	Version
Avaya	
Avaya Aura® Solution for Midsize Enterprise on a HP® Proliant DL360 G7 Server, including: <ul style="list-style-type: none"> • Avaya Aura® Communication Manager • Avaya Aura® Communication Manager Messaging • Avaya Aura® System Manager • Avaya Aura® Session Manager 	6.2.2.0.1120 (System Platform 6.3.0.0.18002) 6.3-03.0.124.0 6.3-26.0 6.3.0 -FP2 6.3.2.0.632028
Avaya Session Border Controller for Enterprise on a Dell R210 V2 Server	6.2.0.Q36
Avaya G450 Media Gateway	33.13.0
Avaya 96x0 Series IP Telephones (H.323)	Avaya one-X Deskphone Edition 3.2
Avaya 96x1 Series IP Telephones (SIP)	Avaya one-X Deskphone Edition SIP 6.2.2.17
Avaya 96x1 Series IP Telephones (H.323)	Avaya one-X Deskphone Edition H.323 6.2.2 (SP2)
Avaya one-X® Communicator (H.323, SIP)	6.1.8.06-SP8-40314
Avaya Flare® Experience for Windows	1.1.3.14
Avaya 9408 Digital Telephone	2.00
Avaya 6210 Analog Telephone	n/a
Alestra Enlace IP	
Sonus Softswitch	V07.03.06 R003
Acme Packet SBC	V6.2
Lucent 5ESS	V16.1

The specific configuration above was used for the compliance testing. Note that this solution will be compatible with other Avaya Server and Media Gateway platforms running similar versions of Communication Manager and Session Manager.

5. Configure Avaya Aura® Communication Manager

This section describes the procedure for configuring Communication Manager for the Alestra Enlace IP SIP trunk service. A SIP trunk is established between Communication Manager and Session Manager for use by signaling traffic to and from Alestra. It is assumed that the general installation of Communication Manager, the Avaya G450 Media Gateway and Session Manager has been previously completed and is not discussed here.

The Communication Manager configuration was performed using the System Access Terminal (SAT). Some screens in this section have been abridged and highlighted for brevity and clarity in presentation.

5.1. Licensing and Capacity

Use the **display system-parameters customer-options** command to verify that the **Maximum Administered SIP Trunks** value on **Page 2** is sufficient to support the desired number of simultaneous SIP calls across all SIP trunks at the enterprise including any trunks to and from the service provider. The example shows that **12000** licenses are available and **263** are in use. The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity.

```
display system-parameters customer-options Page 2 of 11
OPTIONAL FEATURES

IP PORT CAPACITIES USED
Maximum Administered H.323 Trunks: 12000 0
Maximum Concurrently Registered IP Stations: 18000 3
Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
Maximum Concurrently Registered IP eCons: 128 0
Max Concur Registered Unauthenticated H.323 Stations: 100 0
Maximum Video Capable Stations: 36000 0
Maximum Video Capable IP Softphones: 18000 1
Maximum Administered SIP Trunks: 12000 263
Maximum Administered Ad-hoc Video Conferencing Ports: 12000 0
Maximum Number of DS1 Boards with Echo Cancellation: 522 0
Maximum TN2501 VAL Boards: 10 0
Maximum Media Gateway VAL Sources: 250 1
Maximum TN2602 Boards with 80 VoIP Channels: 128 0
Maximum TN2602 Boards with 320 VoIP Channels: 128 0
Maximum Number of Expanded Meet-me Conference Ports: 300 0

(NOTE: You must logoff & login to effect the permission changes.)
```

5.2. System Features

Use the **change system-parameters features** command to set the **Trunk-to-Trunk Transfer** field to **all** to allow incoming calls from the PSTN to be transferred to another PSTN endpoint. If for security reasons, incoming calls should not be allowed to transfer back to the PSTN then leave the field set to **none**.

```
change system-parameters features Page 1 of 20
FEATURE-RELATED SYSTEM PARAMETERS
Self Station Display Enabled? y
Trunk-to-Trunk Transfer: all
Automatic Callback with Called Party Queuing? n
Automatic Callback - No Answer Timeout Interval (rings): 3
Call Park Timeout Interval (minutes): 10
Off-Premises Tone Detect Timeout Interval (seconds): 20
AAR/ARS Dial Tone Required? y

Music (or Silence) on Transferred Trunk Calls? no
DID/Tie/ISDN/SIP Intercept Treatment: attendant
Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
Automatic Circuit Assurance (ACA) Enabled? n

Abbreviated Dial Programming by Assigned Lists? n
Auto Abbreviated/Delayed Transition Interval (rings): 2
Protocol for Caller ID Analog Terminals: Bellcore
Display Calling Number for Room to Room Caller ID Calls? n
```

On **Page 9** verify that a text string has been defined to replace the Calling Party Number (CPN) for restricted or unavailable calls. This text string is entered in the two fields highlighted below. The compliance test used the value of **restricted** for restricted calls and **unavailable** for unavailable calls.

```
change system-parameters features Page 9 of 20
FEATURE-RELATED SYSTEM PARAMETERS

CPN/ANI/ICLID PARAMETERS
CPN/ANI/ICLID Replacement for Restricted Calls: restricted
CPN/ANI/ICLID Replacement for Unavailable Calls: unavailable

DISPLAY TEXT
Identity When Bridging: principal
User Guidance Display? n
Extension only label for Team button on 96xx H.323 terminals? n

INTERNATIONAL CALL ROUTING PARAMETERS
Local Country Code:
International Access Code:
```

5.3. IP Node Names

Use the **change node-names ip** command to verify that node names have been previously defined for the IP addresses of Communication Manager (**procr**) and the Session Manager Security module (**SM**). These node names will be needed for defining the service provider signaling group in **Section 5.6**.

```
change node-names ip Page 1 of 2
```

IP NODE NAMES	
Name	IP Address
Acme_s0p0	192.168.10.53
SM	10.5.5.15
default	0.0.0.0
me-aes	10.5.5.18
procr	10.5.5.12
procr6	::

5.4. Codecs.

Use the **change ip-codec-set** command to define a list of codecs to use for calls between the enterprise and the service provider. For the compliance test, ip-codec-set 2 was used for this purpose. The Alestra Enlace IP SIP Trunk Service preferred codecs are G.729A and G.711A, in this order of preference. Enter **G.729A** and **G.711A** in the **Audio Codec** column of the table. Default values can be used for all other fields.

```
change ip-codec-set 2 Page 1 of 2
```

IP Codec Set			
Codec Set: 2			
Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size(ms)
1: G.729A	<u>n</u>	<u>2</u>	<u>20</u>
2: G.711A	<u>n</u>	<u>2</u>	<u>20</u>
3:			

On **Page 2**, set the **FAX Mode** to **t.38-standard**.

```
change ip-codec-set 2 Page 2 of 2
```

IP Codec Set			
Allow Direct-IP Multimedia? <u>n</u>			
	Mode	Redundancy	ECM: <u>y</u>
FAX	t.38-standard	<u>0</u>	
Modem	<u>off</u>	<u>0</u>	
TDD/TTY	<u>US</u>	<u>3</u>	
Clear-channel	<u>n</u>	<u>0</u>	

5.5. IP Network Regions

Create a separate IP network region for the service provider trunk group. This allows for separate codec or quality of service settings to be used (if necessary) for calls between the enterprise and the service provider versus calls within the enterprise or elsewhere. For the compliance test, IP Network Region 2 was chosen for the service provider trunks. Use the **change ip-network-region 2** command to configure region 2 with the following parameters:

- Set the **Authoritative Domain** field to match the SIP domain of the enterprise. In this configuration, the domain name is **avayalab2.com** as assigned to the shared test environment in the Avaya test lab. This domain name appears in the “From” header of SIP messages originating from this IP region.
- Enter a descriptive name in the **Name** field.
- Leave both **Intra-region** and **Inter-region IP-IP Direct Audio** set to **yes**, the default setting. This will enable **IP-IP Direct Audio** (shuffling), to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway. Shuffling can be further restricted at the trunk level on the Signaling Group form.
- Set the **Codec Set** field to the IP codec set defined in **Section 5.4**.
- Default values can be used for all other fields.

```
change ip-network-region 2                                     Page 1 of 20
                                                                IP NETWORK REGION
Region: 2
Location: 1          Authoritative Domain: avayalab2.com
Name: Alestra SIP Trunk      Stub Network Region: n
MEDIA PARAMETERS
Codec Set: 2          Intra-region IP-IP Direct Audio: yes
                    Inter-region IP-IP Direct Audio: yes
                    IP Audio Hairpinning? n
UDP Port Min: 2048
UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
Call Control PHB Value: 46
Audio PHB Value: 46
Video PHB Value: 26
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 6
Audio 802.1p Priority: 6
Video 802.1p Priority: 5
AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS
H.323 Link Bounce Recovery? n
Idle Traffic Interval (sec): 20
Keep-Alive Interval (sec): 5
Keep-Alive Count: 5
                    RSUP Enabled? n
```

On **Page 4**, define the IP codec set to be used for traffic between region 2 and region 1 (the rest of the enterprise). Enter the desired IP codec set in the **codec set** column of the row with destination region (**dst rgn**) 1. Default values may be used for all other fields. The following example shows the settings used for the compliance test. It indicates that codec set **2** will be used for calls between region 2 (the service provider region) and region 1 (the rest of the enterprise).

change ip-network-region 2										Page 4 of 20		
Source Region: 2		Inter Network Region Connection Management							I	M		
dst rgn	codec set	direct WAN	WAN-BW-limits Units	Video Total Norm	Intervening Prio Shr	Regions	Dyn CAC	A R	G L	t c e t		
1	2	y	NoLimit				n			t		
2	2								all			
3												
4												

5.6. Signaling Group

Use the **add signaling-group** command to create a signaling group between Communication Manager and Session Manager for use by the service provider trunk. This signaling group is used for inbound and outbound calls between the service provider and the enterprise. For the compliance test, signaling group 2 was used for this purpose and was configured using the parameters highlighted below:

- Set the **Group Type** field to **sip**.
- Set the **IMS Enabled** field to **n**. This specifies the Communication Manager will serve as an Evolution Server for the Session Manager.
- Set the **Transport Method** to the transport protocol to be used between Communication Manager and Session Manager. For the compliance test, **tcp** was used.
- Set the **Peer Detection Enabled** field to **y**. The **Peer-Server** field will initially be set to **Others** and cannot be changed via administration. Later, the **Peer-Server** field will automatically change to **SM** once Communication Manager detects its peer is a Session Manager.

Note: Once the **Peer-Server** field is updated with **SM**, the system changes the default values of the following fields, setting them to display-only:

- **Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers?** is changed to **y**.
- **Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers?** is changed to **n**.
- Set the **Near-end Node Name** to **procr**. This node name maps to the IP address of the Communication Manager as defined in **Section 5.3**.
- Set the **Far-end Node Name** to **SM**. This node name maps to the IP address of Session Manager, as defined in **Section 5.3**.

```

change signaling-group 2                                     Page 1 of 2
                    SIGNALING GROUP

Group Number: 2                                           Group Type: sip
IMS Enabled? n                                           Transport Method: tcp
  Q-SIP? n
  IP Video? n                                           Enforce SIPS URI for SRTP? y
Peer Detection Enabled? y Peer Server: SM
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n

Near-end Node Name: procr                               Far-end Node Name: SM
Near-end Listen Port: 5062                             Far-end Listen Port: 5062
                                                    Far-end Network Region: 2

Far-end Domain: avayalab2.com

Incoming Dialog Loopbacks: eliminate                    Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload                               RFC 3389 Comfort Noise? n
Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
Enable Layer 3 Test? y                                 Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n                 Alternate Route Timer(sec): 6

```

- Set the **Near-end Listen Port** and **Far-end Listen Port** to a valid unused port instead of the default well-known port value. (For TCP, the well-known port value is 5060). This is necessary so the SM can distinguish this trunk from the trunk used for other enterprise SIP traffic. For the compliance test both the **Near-end Listen Port** and **Far-end Listen Port** were set to **5062**.
- Set the **Far-end Network Region** to the IP network region defined for the service provider in **Section 5.5**.
- Set the **Far-end Domain** to the domain of the enterprise.
- Set the **DTMF over IP** field to **rtp-payload**. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- Set **Direct IP-IP Audio Connections** to **y**. This field will enable media shuffling on the SIP trunk allowing Communication Manager to redirect media traffic directly between the SIP trunk and the enterprise endpoint. If this value is set to **n**, then the Avaya Media Gateway will remain in the media path of all calls between the SIP trunk and the endpoint. Depending on the number of media resources available in the Avaya Media Gateway, these resources may be depleted during high call volume preventing additional calls from completing.
- Default values may be used for all other fields.

5.7. Trunk Group

Use the **add trunk-group** command to create a trunk group for the signaling group created in **Section 5.6**. For the compliance test, trunk group 2 was configured using the parameters highlighted below.

- Set the **Group Type** field to *sip*.
- Enter a descriptive name for the **Group Name**.
- Enter an available trunk access code (TAC) that is consistent with the existing dial plan in the **TAC** field.
- Set the **Service Type** field to *public-ntwrk*.
- Set the **Signaling Group** to the signaling group shown in the previous step.
- Set the **Number of Members** field to the number of trunk members in the SIP trunk group. This value determines how many simultaneous SIP calls can be supported by this trunk.
- Default values were used for all other fields.

```
change trunk-group 2                                     Page 1 of 21
                                     TRUNK GROUP
Group Number: 2          Group Type: sip          CDR Reports: y
Group Name: Service Provider          COR: 1          TN: 1          TAC: 602
Direction: two-way          Outgoing Display? n
Dial Access? n          Night Service: _____
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n
                                     Member Assignment Method: auto
                                     Signaling Group: 2
                                     Number of Members: 6
```

On **Page 2**, verify that the **Preferred Minimum Session Refresh Interval** is set to a value acceptable to the service provider. This value defines the interval that re-INVITEs must be sent to keep the active session alive. For the compliance test, the default value of **600** seconds was used.

```
change trunk-group 2                                     Page 2 of 21
Group Type: sip
TRUNK PARAMETERS
Unicode Name: auto
Redirect On OPTIM Failure: 5000
SCCAN? n          Digital Loss Group: 18
Preferred Minimum Session Refresh Interval(sec): 600
Disconnect Supervision - In? y Out? y
```

On **Page 3**, set the **Numbering Format** field to *private*. This field specifies the format of the calling party number (CPN) sent to the far-end. Beginning with Communication Manager 6.0, public numbers are automatically preceded with a + sign when passed in the SIP From, Contact and P-Asserted Identity headers. The addition of the + sign impacted interoperability with Alestra. Thus, the **Numbering Format** was set to **private** and the **Numbering Format** in the route pattern was set to **unk-unk** (see **Section 5.10**). Set the **Replace Restricted Numbers** and **Replace Unavailable Numbers** fields to **y**. This will allow the CPN displayed on local endpoints to be replaced with the value set in **Section 5.2**, if the inbound call enabled CPN block.

```
change trunk-group 2                                     Page 3 of 21
TRUNK FEATURES
  ACA Assignment? n                                     Measured: none
                                                    Maintenance Tests? y

  Numbering Format: private
                                                    UII Treatment: service-provider
                                                    Replace Restricted Numbers? y
                                                    Replace Unavailable Numbers? y
```

On **Page 4**, leave the **Network Call Redirection** and the **Send Diversion Header** fields set to the default value **n**. Alestra does not support the use of the SIP REFER method or the Diversion header for calls that are transferred or forwarded back to the PSTN. Set the **Support Request History** field to **n**. Set the **Telephone Event Payload Type** to **101**, and **Convert 180 to 183 for Early Media** to **y**, the values preferred by Alestra. Default values were used for all other fields.

```
change trunk-group 2                                     Page 4 of 21
PROTOCOL VARIATIONS
  Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
  Send Transferring Party Information? n
  Network Call Redirection? n
  Send Diversion Header? n
  Support Request History? n
  Telephone Event Payload Type: 101
  Convert 180 to 183 for Early Media? y
  Always Use re-INVITE for Display Updates? n
  Identity for Calling Party Display: P-Asserted-Identity
  Block Sending Calling Party Location in INVITE? n
  Accept Redirect to Blank User Destination? n
  Enable Q-SIP? n
```

5.8. Calling Party Information

The calling party number is sent in the SIP “From”, “Contact” and “PAI” headers. Since private numbering was selected to define the format of this number (**Section 5.7**), use the **change private-numbering** command to create an entry for each extension which has a DID assigned. DID numbers are provided by the SIP service provider. Each DID number is assigned in this table to one enterprise internal extension or Vector Directory Numbers (VDNs), and they are used to authenticate the caller. In the sample configuration, 6 DID numbers were assigned for testing. The example below shows these 6 numbers, mapped to 6 extensions used during testing. These 10-digit numbers were used in the outbound calling party information on the service provider trunk when calls were originated from these extensions.

```
change private-numbering 1 Page 1 of 2
```

NUMBERING - PRIVATE FORMAT				
Ext Len	Ext Code	Trk Grp(s)	Private Prefix	Total Len
5	5	3		5
5	55001	2	8111111234	10
5	55002	2	8111111235	10
5	55003	2	8111111236	10
5	55004	2	8111111237	10
5	55005	2	8111111238	10
5	55006	2	8111111239	10

Total Administered: 17
Maximum Entries: 540

5.9. Inbound Routing

In general, the “incoming call handling treatment” form for a trunk group can be used to manipulate the digits received for an incoming call if necessary. Since Session Manager is present, Session Manager can be used to perform digit conversion using an Adaptation, and digit manipulation via the Communication Manager incoming call handling table may not be necessary. If the DID number sent by Alestra is left unchanged by Session Manager, then the DID number can be mapped to an extension using the incoming call handling treatment of the receiving trunk group. During the compliance test, Alestra sent to the enterprise the last 4 digits of the DID number. Use the **change inc-call-handling-trmt** command to create an entry for each DID.

```
change inc-call-handling-trmt trunk-group 2 Page 1 of 30
```

INCOMING CALL HANDLING TREATMENT				
Service/ Feature	Number Len	Number Digits	Del	Insert
public-ntwrk	4	1234	4	55001
public-ntwrk	4	1235	4	55002
public-ntwrk	4	1236	4	55003
public-ntwrk	4	1237	4	55004
public-ntwrk	4	1238	4	55005
public-ntwrk	4	1239	4	55006
public-ntwrk				

5.10. Outbound Routing

In these Application Notes, the Automatic Route Selection (ARS) feature is used to route outbound calls via the SIP trunk to the service provider. In the sample configuration, the single digit 9 is used as the ARS access code. Enterprise callers will dial 9 to reach an “outside line”. This common configuration is illustrated below with little elaboration. Use the **change dialplan analysis** command to define a dialed string beginning with **9** of length **1**, as a feature access code (**fac**).

```
change dialplan analysis Page 1 of 12
```

DIAL PLAN ANALYSIS TABLE								
Location: all						Percent Full: 2		
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type
0	1	attd						
1	5	ext						
2	5	ext						
3	4	ext						
4	5	ext						
5	5	ext						
6	3	dac						
7	5	ext						
8	5	ext						
9	1	fac						
*	3	dac						
#	3	dac						

Use the **change feature-access-codes** command to configure **9** as the **Auto Route Selection (ARS) – Access Code 1**.

```
change feature-access-codes Page 1 of 10
```

FEATURE ACCESS CODE (FAC)		
Abbreviated Dialing List1 Access Code:	*10	
Abbreviated Dialing List2 Access Code:	*12	
Abbreviated Dialing List3 Access Code:	*13	
Abbreviated Dial - Prgm Group List Access Code:	*14	
Announcement Access Code:	*19	
Answer Back Access Code:	_____	
Auto Alternate Routing (AAR) Access Code:	*00	
Auto Route Selection (ARS) - Access Code 1:	9	Access Code 2: _____
Automatic Callback Activation:	*33	Deactivation: #33
Call Forwarding Activation Busy/DA:	*30	All: *31
Call Forwarding Enhanced Status:	_____	Act: _____
		Deactivation: _____

Use the **change ars analysis** command to configure the routing of dialed digits following the first digit 9. The example below shows a subset of the dialed strings tested as part of the compliance test. See **Section 2.1** for the complete list of call types tested. All dialed strings are mapped to route pattern 2 which contains the SIP trunk group to the service provider.

```
change ars analysis 0 Page 1 of 2
```

ARS DIGIT ANALYSIS TABLE							
Location: all							
Percent Full: 1							
Dialed String	Total		Route	Call	Node	ANI	
	Min	Max	Pattern	Type	Num	Reqd	
00	12	22	2	intl		n	
001	13	13	2	intl		n	
01	12	12	2	natl		n	
040	3	3	2	svcl		n	
2	8	8	2	hnpa		n	
8	8	8	2	hnpa		n	

The route pattern defines which trunk group will be used for the call and performs any necessary digit manipulation. Use the **change route-pattern** command to configure the parameters for the service provider trunk route pattern in the following manner. The example below shows the values used for route pattern 2 for the compliance test.

- **Pattern Name:** Enter a descriptive name.
- **Grp No:** Enter the outbound trunk group for the SIP service provider. For the compliance test, trunk group 2 was used.
- **FRL:** Set the Facility Restriction Level (FRL) field to a level that allows access to this trunk for all users that require it. The value of 0 is the least restrictive level.
- **Numbering Format:** Set to *unk-unk*. All calls using this route pattern will use the private numbering table. See setting of the **Numbering Format** in the trunk group form for full details in **Section 5.7**.
- Default values were used for all other fields.

```
change route-pattern 2 Page 1 of 3
```

Pattern Number: 2												Pattern Name: Alestra SIP Trk	
SCCAN? n												Secure SIP? n	
Grp No	FRL	NPA	Pfx Mrk	Hop Lmt	Toll List	No. Del	Inserted Dgts					DCS/ QSIG Intw	IXC
1:	2	0										n	user
2:												n	user
3:												n	user
4:												n	user
5:												n	user
6:												n	user

BCC VALUE		TSC	CA-TSC	ITC		BCIE	Service/Feature		PARM	No. Dgts	Numbering Format	LAR
0	1 2 M 4 W		Request									
1:	y y y y y n	n				rest					unk-unk	none
2:	u u u u u n	n				rest						none

6. Configure Avaya Aura® Session Manager

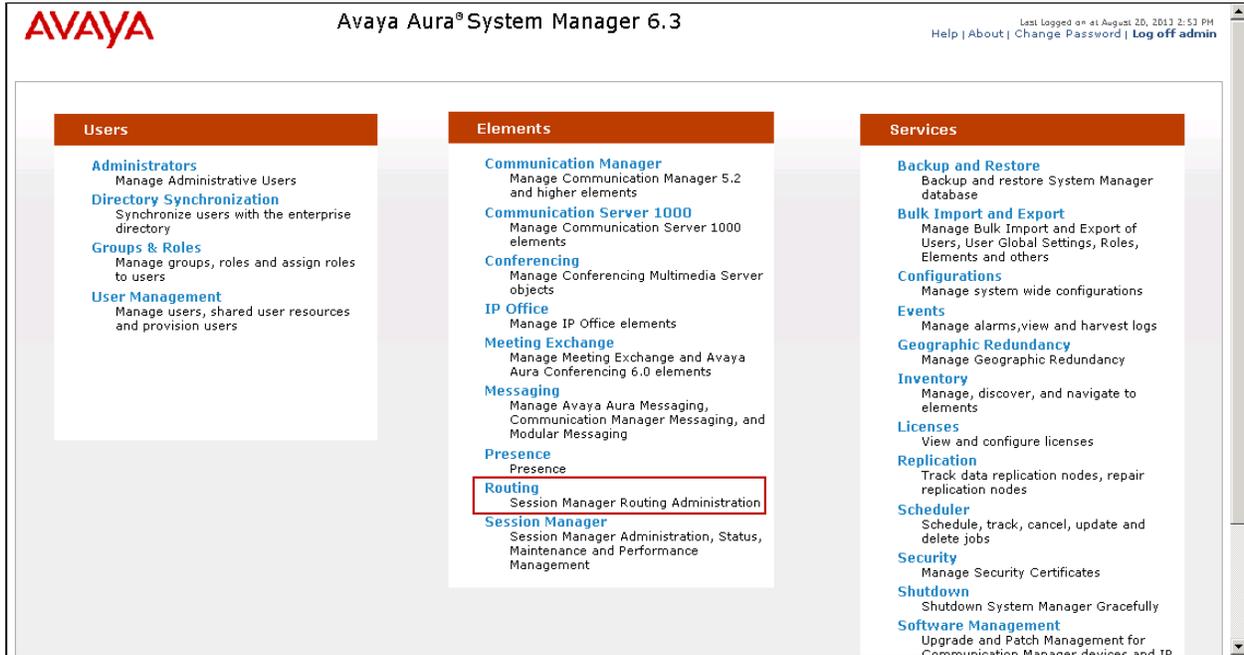
This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain
- Logical/physical Location that can be occupied by SIP Entities
- SIP Entities corresponding to Communication Manager, Session Manager and the Avaya SBCE
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Routing Policies, which control call routing between the SIP Entities
- Dial Patterns, which govern to which SIP Entity a call is routed
- Session Manager instance, corresponding to the Session Manager Server to be managed by System Manager

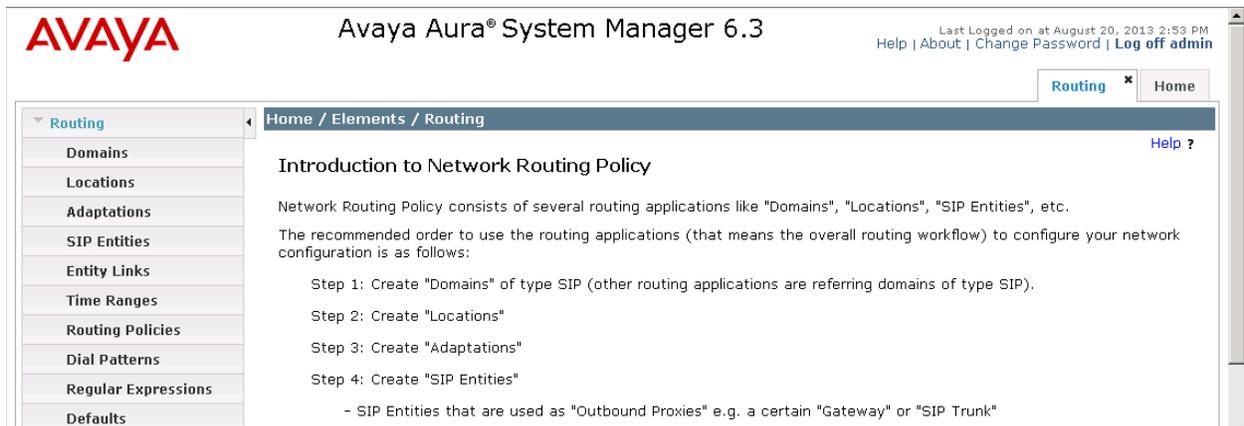
It may not be necessary to create all the items above when creating a connection to the service provider, since some of them would have already been defined as part of the initial Midsize Enterprise Solution template installation. This includes entries such as certain SIP domains, locations, SIP entities, and Session Manager itself. However, each item should be reviewed to verify the configuration.

6.1. System Manager Login and Navigation

Session Manager configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL “https://<ip-address>/SMGR”, where “<ip-address>” is the IP address of System Manager. Log in with the appropriate credentials and click on **Log On** (not shown). The screen shown below is then displayed; click on **Routing**.



The navigation tree displayed in the left pane below will be referenced in subsequent sections to navigate to items requiring configuration. Most items discussed in this section will be located under the **Routing** link shown below.



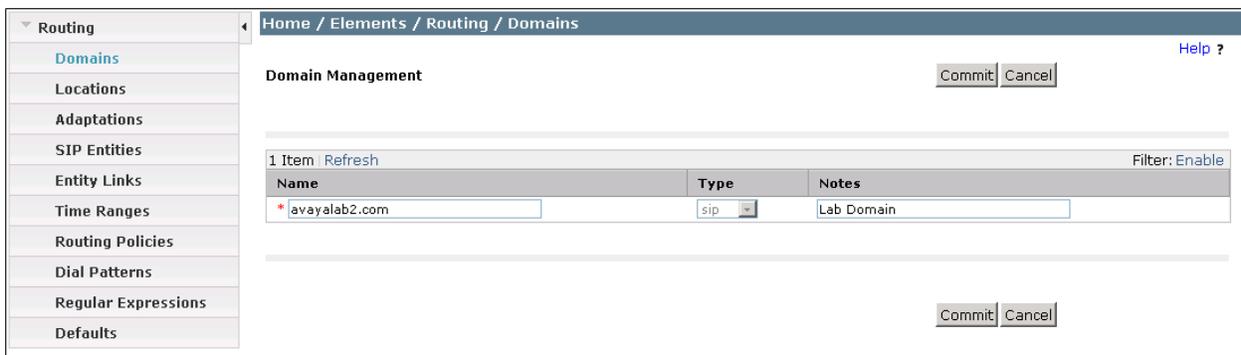
6.2. SIP Domain

Create a SIP domain for each domain for which Session Manager will need to be aware in order to route calls. For the compliance test, this will be the enterprise domain, **avayalab2.com**.

Navigate to **Routing → Domains** in the left-hand navigation pane (**Section 6.1**) and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

- **Name:** Enter the domain name.
- **Type:** Select **sip** from the pull-down menu.
- **Notes:** Add a brief description (optional).

Click **Commit**. The screen below shows the entry for the enterprise domain



The screenshot shows the 'Domain Management' interface. The breadcrumb is 'Home / Elements / Routing / Domains'. The page title is 'Domain Management' with 'Commit' and 'Cancel' buttons. Below the title is a table with one item:

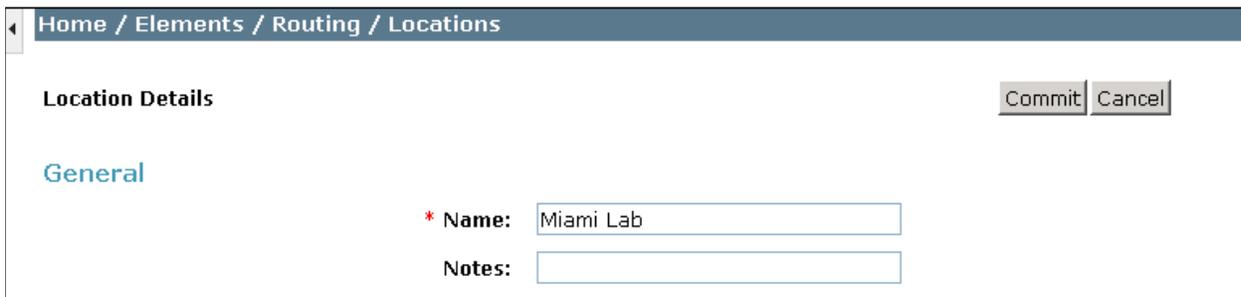
Name	Type	Notes
* avayalab2.com	sip	Lab Domain

At the bottom of the table, there are 'Commit' and 'Cancel' buttons. A 'Filter: Enable' link is also visible.

6.3. Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management, call admission control and location-based routing. For the compliance test, only one location was defined, named **Miami Lab**, which contains all the equipment at the enterprise. To add a location, navigate to **Routing → Locations** in the left-hand navigation pane and click the **New** button in the right pane (not shown). In the **General** section, enter the following values.

- **Name:** Enter a descriptive name for the location.
- **Notes:** Add a brief description (optional).



The screenshot shows the 'Location Details' interface. The breadcrumb is 'Home / Elements / Routing / Locations'. The page title is 'Location Details' with 'Commit' and 'Cancel' buttons. Below the title is the 'General' section with the following fields:

* **Name:** Miami Lab

Notes:

In the **Location Pattern** section, click **Add** and enter the following values. Use default values for all remaining fields:

- **IP Address Pattern:** Add IP address patterns used to identify the location.
- **Notes:** Add a brief description (optional).

Click **Commit** to save.

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	*10.5.5.*	CM/SM/ASBCE
<input type="checkbox"/>	*192.168.10.*	Phones/Others

Select : All, None

Commit Cancel

6.4. SIP Entities

A SIP Entity must be added for Session Manager and for each SIP telephony system connected to it, which includes Communication Manager and the Avaya SBCE. Navigate to **Routing** → **SIP Entities** in the left navigation pane and click on the **New** button in the right pane (not shown). In the **General** section, enter the following values. Use default values for all remaining fields:

- **Name:** Enter a descriptive name.
- **FQDN or IP Address:** Enter the FQDN or IP address of the SIP Entity that is used for SIP signaling.
- **Type:** Select **Session Manager** for Session Manager, **CM** for Communication Manager and **SIP Trunk** for the Avaya SBCE
- **Adaptation:** This field is only present if **Type** is not set to **Session Manager**. If Adaptations were to be created, here is where they would be applied to the entity.
- **Location:** Select the location that applies to the SIP Entity being created. For the compliance test, all components were located in location **Miami Lab**.
- **Time Zone:** Select the time zone for the location above.

The following screen shows the addition of the Session Manager SIP Entity. The IP address of the Session Manager Security Module is entered for **FQDN or IP Address**.

Home / Elements / Routing / SIP Entities

SIP Entity Details Commit Cancel

General

* **Name:**

* **FQDN or IP Address:**

Type:

Notes:

Location:

Outbound Proxy:

Time Zone:

Credential name:

SIP Link Monitoring

SIP Link Monitoring:

To define the ports that Session Manager will use to listen for SIP requests, scroll down to the **Port** section of the **SIP Entity Details** screen. This section is only present for **Session Manager** SIP entities. The screen below shows the ports used by Session Manager in the shared lab environment. Only TCP ports 5060 and 5062 are directly relevant to these Application Notes.

Port

TCP Failover port:

TLS Failover port:

Add Remove

4 Items Refresh Filter: Enable

<input type="checkbox"/>	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	<input type="text" value="5060"/>	TCP	avayalab2.com	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="5061"/>	TLS	avayalab2.com	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="5062"/>	TCP	avayalab2.com	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="6060"/>	TCP	avayalab2.com	<input type="text"/>

Select : All, None

The following screen shows the addition of the SIP Entity for Communication Manager. In order for Session Manager to send SIP service provider traffic on a separate entity link to Communication Manager, the creation of a separate SIP entity for Communication Manager is required. This SIP Entity should be different to the one created during the Template installation to be used by all other enterprise SIP traffic. The **FQDN or IP Address** field is set to the IP address of the “**procr**” interface in Communication Manager.

Home / Elements / Routing / SIP Entities

SIP Entity Details Commit Cancel

General

* Name: C.M. Trunk 2

* FQDN or IP Address: 10.5.5.12

Type: CM

Notes:

Adaptation:

Location: Miami Lab

Time Zone: America/New_York

Override Port & Transport with DNS SRV:

* SIP Timer B/F (in seconds): 4

Credential name:

Call Detail Recording: none

Loop Detection

Loop Detection Mode: Off

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

The following screen shows the addition of the Avaya SBCE Entity. The **FQDN or IP Address** field is set to the IP address of the SBC private network interface (see **Figure 1**).

Home / Elements / Routing / SIP Entities

SIP Entity Details Commit Cancel

General

* **Name:** Avaya SBCE

* **FQDN or IP Address:** 10.5.5.72

Type: SIP Trunk

Notes:

Adaptation:

Location: Miami Lab

Time Zone: America/New_York

Override Port & Transport with DNS SRV:

* **SIP Timer B/F (in seconds):** 4

Credential name:

Call Detail Recording: egress

Loop Detection

Loop Detection Mode: Off

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

6.5. Entity Links

A SIP trunk between Session Manager and a telephony system is described by an Entity Link. Two Entity Links were created; one to the Communication Manager for use only by service provider traffic and one to the Avaya SBCE. To add an Entity Link, navigate to **Routing** → **Entity Links** in the left navigation pane and click on the **New** button in the right pane (not shown). Fill in the following fields in the new row that is displayed:

- **Name:** Enter a descriptive name.
- **SIP Entity 1:** Select the Session Manager from the drop-down menu.
- **Protocol:** Select the transport protocol used for this link.
- **Port:** Port number on which Session Manager will receive SIP requests from the far-end.
- **SIP Entity 2:** Select the name of the other system from the drop-down menu.
- **Port:** Port number on which the other system receives SIP requests from the Session Manager.
- **Connection Policy:** Select **Trusted** to allow calls from the associated SIP Entity.

Click **Commit** to save.

The screen below shows the Entity Link to Communication Manager. The protocol and ports defined here must match the values used on the Communication Manager signaling group form in **Section 5.6**.

The screenshot shows the 'Entity Links' configuration page. The breadcrumb is 'Home / Elements / Routing / Entity Links'. There are 'Commit' and 'Cancel' buttons. Below the title, there is a table with one row of configuration data. The table has columns for Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, and Connection Policy. The values in the row are: Name: SM to CM Trunk 2, SIP Entity 1: SessionManager1, Protocol: TCP, Port: 5062, SIP Entity 2: C.M. Trunk 2, Port: 5062, Connection Policy: trusted.

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy
<input type="checkbox"/>	* SM to CM Trunk 2	* SessionManager1	TCP	* 5062	* C.M. Trunk 2	* 5062	trusted

Entity Link to the Avaya SBCE:

The screenshot shows the 'Entity Links' configuration page. The breadcrumb is 'Home / Elements / Routing / Entity Links'. There are 'Commit' and 'Cancel' buttons. Below the title, there is a table with one row of configuration data. The table has columns for Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, and Connection Policy. The values in the row are: Name: SM to ASBCE, SIP Entity 1: SessionManager1, Protocol: TCP, Port: 5060, SIP Entity 2: Avaya SBCE, Port: 5060, Connection Policy: trusted.

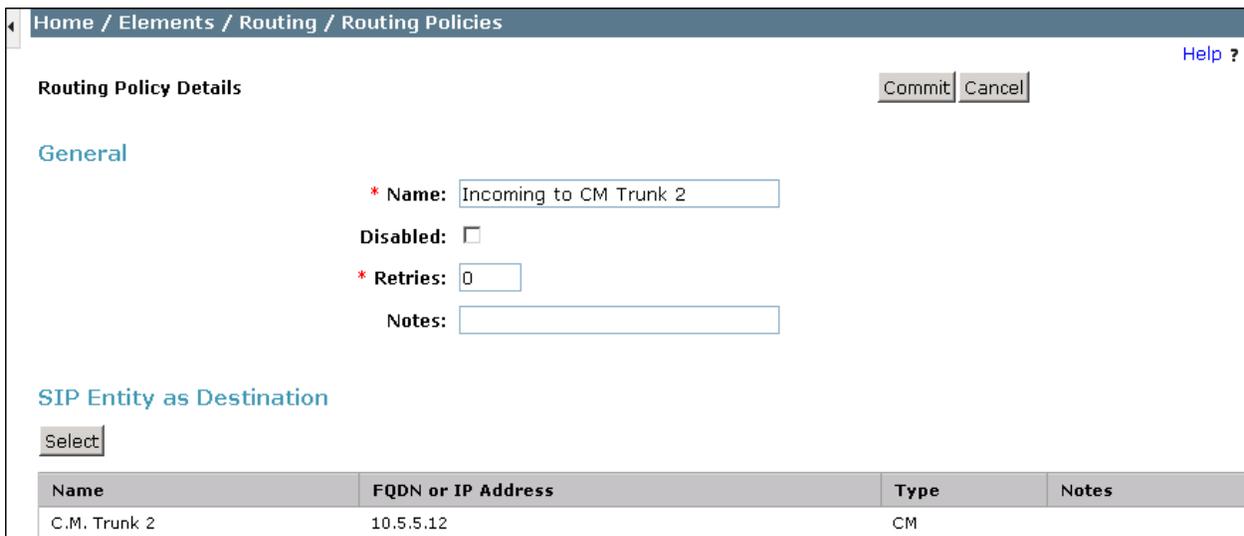
<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy
<input type="checkbox"/>	* SM to ASBCE	* SessionManager1	TCP	* 5060	* Avaya SBCE	* 5060	trusted

6.6. Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.4**. Two routing policies must be added: one for Communication Manager and one for the Avaya SBCE. To add a routing policy, navigate to **Routing → Routing Policies** in the left navigation pane and click on the **New** button in the right pane (not shown). The following screen is displayed. In the **General** section, enter a descriptive **Name** and add a brief description under **Notes** (optional).

In the **SIP Entity as Destination** section, click **Select**. The **SIP Entity List** page opens (not shown). Select the appropriate SIP entity to which this routing policy applies and click **Select**. The selected SIP Entity displays on the **Routing Policy Details** page as shown below. Use default values for remaining fields. Click **Commit** to save.

The following screens show the Routing Policies for Communication Manager and the Avaya SBCE



Home / Elements / Routing / Routing Policies Help ?

Routing Policy Details Commit Cancel

General

* Name:

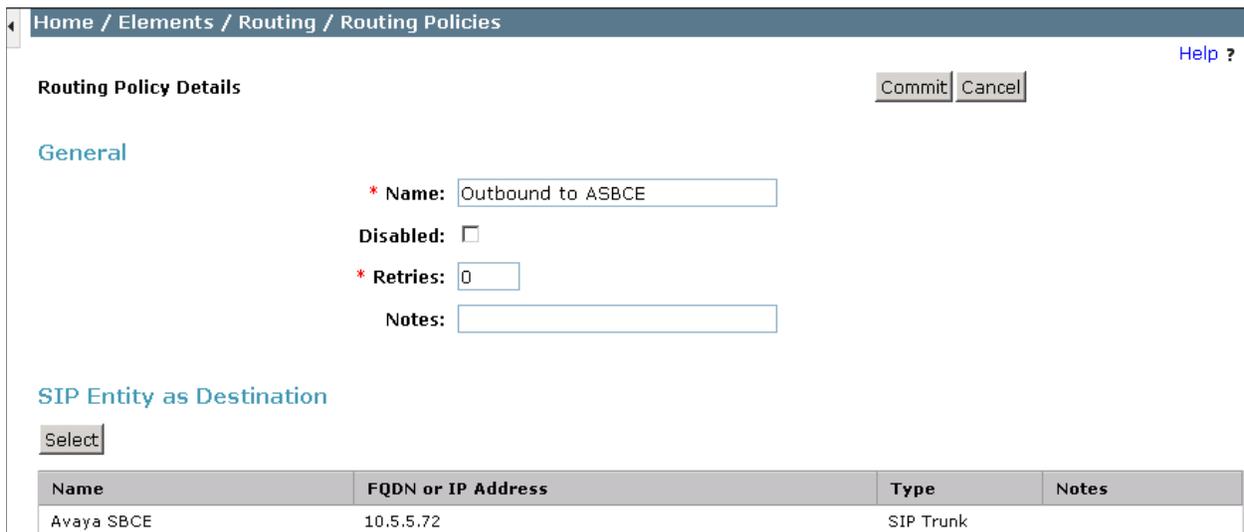
Disabled:

* Retries:

Notes:

SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
C.M. Trunk 2	10.5.5.12	CM	



Home / Elements / Routing / Routing Policies Help ?

Routing Policy Details Commit Cancel

General

* Name:

Disabled:

* Retries:

Notes:

SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
Avaya SBCE	10.5.5.72	SIP Trunk	

6.7. Dial Patterns

Dial Patterns are needed to route specific calls through Session Manager. For the compliance test, dial patterns were needed to route calls from Communication Manager to Alestra and vice versa. Dial Patterns define which route policy will be selected for a particular call based on the dialed digits, destination domain and originating location. To add a dial pattern, navigate to **Routing → Dial Patterns** in the left navigation pane and click on the **New** button in the right pane (not shown). Fill in the following, as shown in the screens below:

In the **General** section, enter the following values:

- **Pattern:** Enter a dial string that will be matched against the Request-URI of the call.
- **Min:** Enter a minimum length used in the match criteria.
- **Max:** Enter a maximum length used in the match criteria.
- **SIP Domain:** Enter the destination domain used in the match criteria.
- **Notes:** Add a brief description (optional).

In the **Originating Locations and Routing Policies** section, click **Add**. From the **Originating Locations and Routing Policy List** that appears (not shown), select the appropriate originating location for use in the match criteria. Lastly, select the routing policy from the list that will be used to route all calls that match the specified criteria. Click **Select**.

Default values can be used for the remaining fields. Click **Commit** to save.

Two examples of the Dial Patterns used for the compliance test are shown, one for outbound calls from the enterprise to the PSTN and one for inbound calls. Other Dial Patterns (e.g., 01 for long distance national, 001 for international calls to the U.S., etc.) were similarly defined.

The example in this screen shows that in the test environment, 8 digit dialed numbers for outbound local calls in Monterrey, Mexico, beginning with 8 and originating from the Miami Lab location uses route policy **Outbound to ASBCE**, which sends the call out to the PSTN via the Avaya SBCE to the Alestra SIP Trunk.

Home / Elements / Routing / Dial Patterns Help ?

Dial Pattern Details Commit Cancel

General

* Pattern:

* Min:

* Max:

Emergency Call:

Emergency Priority:

Emergency Type:

SIP Domain:

Notes:

Originating Locations and Routing Policies

Add Remove

1 Item Refresh Filter: Enable

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Miami Lab		Outbound to ASBCE	0	<input type="checkbox"/>	Avaya SBCE	

Repeat this procedure as needed, to define additional dial patterns for PSTN numbers to be routed to the Alestra network via the Avaya SBCE.

On incoming calls, Alestra sent to the enterprise during the compliance test only the last 4 digits of the DID number. The example below shows that a 4 digit number starting with **1**, which is the DID range assigned by Alestra, will use route policy **Incoming to CM Trunk 2** to Communication Manager.

Home / Elements / Routing / Dial Patterns Help ?

Dial Pattern Details Commit Cancel

General

* Pattern:

* Min:

* Max:

Emergency Call:

Emergency Priority:

Emergency Type:

SIP Domain:

Notes:

Originating Locations and Routing Policies

Add Remove

1 Item Refresh Filter: Enable

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Miami Lab		Incoming to CM Trunk 2		<input type="checkbox"/>	C.M. Trunk 2	

Repeat this procedure as needed to define additional dial patterns for other numbers assigned to the enterprise by Alestra, to be routed to Communication Manager.

6.8. Add/View Session Manager Instance

The creation of a Session Manager Instance provides the linkage between System Manager and Session Manager. This was most likely done as part of the initial Session Manager installation. To add a Session Manager, navigate to **Elements → Session Manager → Session Manager Administration** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). If the Session Manager already exists, click **View** (not shown) to view the configuration. Enter/verify the data as described below and shown in the following screen:

In the **General** section, enter the following values:

- **SIP Entity Name:** Select the SIP Entity created for Session Manager.
- **Description:** Add a brief description (optional).
- **Management Access Point Host Name/IP:** Enter the IP address of the Session Manager management interface.

The screen below shows the Session Manager values used for the compliance test.

The screenshot shows the 'View Session Manager' configuration page. The left sidebar contains a navigation menu with items like Dashboard, Session Manager Administration, Communication Profile Editor, Network Configuration, Device and Location Configuration, Application Configuration, System Status, System Tools, and Performance. The main content area is titled 'View Session Manager' and includes a breadcrumb trail: Home / Elements / Session Manager / Session Manager Administration. Below the title, there are links for 'General | Security Module | NIC Bonding | Monitoring | CDR | Personal Profile Manager (PPM) - Connection Settings | Event Server | Expand All | Collapse All'. The 'General' tab is selected, showing the following configuration fields:

- SIP Entity Name:** SessionManager1
- Description:** (empty)
- Management Access Point Host Name/IP:** 10.5.5.14
- Direct Routing to Endpoints:** Enable
- VMware Virtual Machine:**

In the **Security Module** section, enter the following values:

- **SIP Entity IP Address:** Should be filled in automatically based on the SIP Entity Name. Otherwise, enter IP address of Session Manager signaling interface.
- **Network Mask:** Enter the network mask corresponding to the IP address of Session Manager.
- **Default Gateway:** Enter the IP address of the default gateway for Session Manager.

Use default values for the remaining fields. Click **Save** (not shown) to add this Session Manager. The screen below shows Security Module values used for the compliance test.

The screenshot shows the 'Security Module' configuration page. The 'Security Module' tab is selected, displaying the following configuration fields:

- SIP Entity IP Address:** 10.5.5.15
- Network Mask:** 255.255.255.0
- Default Gateway:** 10.5.5.254
- Call Control PHB:** 46
- QOS Priority:** 6
- Speed & Duplex:** Auto
- VLAN ID:** (empty)

7. Configure Avaya Session Border Controller for Enterprise

In the sample configuration, the Avaya SBCE is used as the edge device between the Avaya CPE and the Alestra Enlace IP SIP Trunk Service. It is assumed that the initial installation of the Avaya SBCE and the assignment of the management interface IP Address have already been completed; hence these tasks are not covered in these Application Notes. For more information on the installation and initial provisioning of the Avaya SBCE consult [10] and [11] in the **References** section.

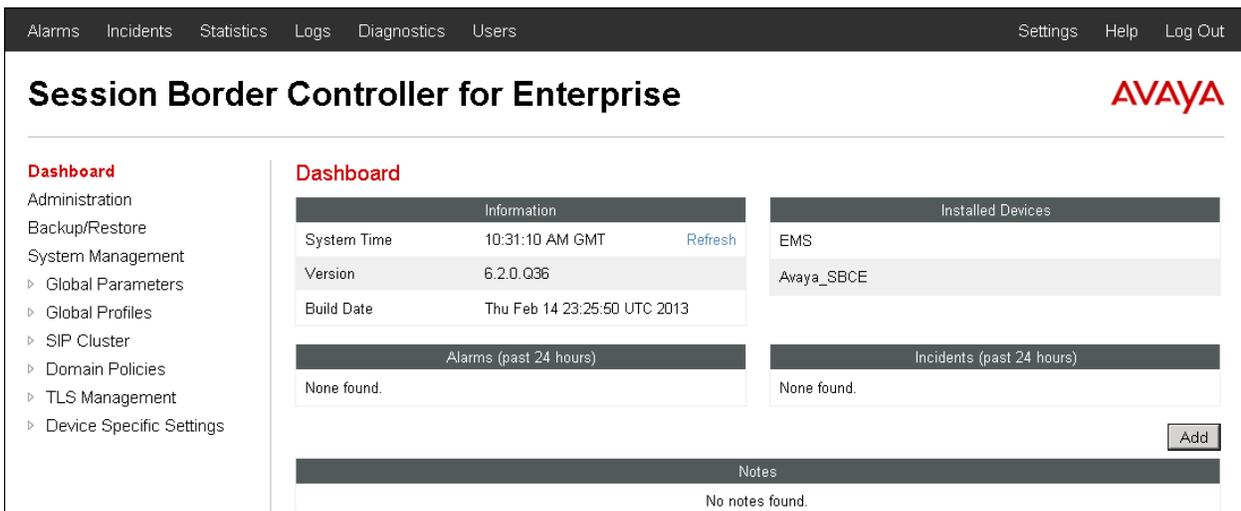
7.1. System Access

Access the Session Border Controller web management interface by using a web browser and entering the URL **https://<ip-address>**, where **<ip-address>** is the management IP address configured at installation. Log in using the appropriate credentials.



The screenshot shows the login page for the Avaya Session Border Controller for Enterprise. On the left is the Avaya logo and the product name. On the right is a 'Log In' section with input fields for 'Username:' and 'Password:', and a 'Log In' button. Below the button is a disclaimer: 'This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use or modifications of this system is strictly prohibited. Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal or other applicable domestic and foreign laws. The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials. All users must comply with all corporate instructions regarding the protection of information assets. © 2011 - 2013 Avaya Inc. All rights reserved.'

Once logged in, the Dashboard screen is presented. The left navigation pane contains the different available menu items used for the configuration of the Avaya SBCE.



The screenshot shows the dashboard of the Avaya Session Border Controller for Enterprise. At the top is a navigation bar with links for Alarms, Incidents, Statistics, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main content area is titled 'Session Border Controller for Enterprise' and features a left-hand navigation menu with categories like Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, SIP Cluster, Domain Policies, TLS Management, and Device Specific Settings. The main dashboard area contains several widgets: 'Information' with system time (10:31:10 AM GMT), version (6.2.0.Q36), and build date (Thu Feb 14 23:25:50 UTC 2013); 'Installed Devices' showing one device named 'Avaya_SBCE'; 'Alarms (past 24 hours)' and 'Incidents (past 24 hours)' both showing 'None found.'; and 'Notes' showing 'No notes found.' with an 'Add' button.

7.2. System Management

To view current system information, select **System Management** on the left navigation pane. A list of installed devices is shown in the right pane. In the reference configuration, a single device named **Avaya_SBCE** is shown. The management IP address that was configured during installation and the current software version are shown here. Note that the management IP address needs to be on a subnet separate from the ones used in the other interfaces of the Avaya SBCE, segmented from all VoIP traffic. Verify the device shows the status of **Commissioned**, indicating that the initial installation process of the device has been previously completed, as shown on the screen below.

Session Border Controller for Enterprise AVAYA

Dashboard
Administration
Backup/Restore
System Management
‣ Global Parameters
‣ Global Profiles
‣ SIP Cluster
‣ Domain Policies

System Management

Devices Updates SSL VPN Licensing

Device Name (Serial Number)	Management IP	Version	Status	
Avaya_SBCE (9P831020132)	192.168.10.70	6.2.0.Q36	Commissioned	Reboot Shutdown Restart Application View Edit Delete

To view the network information assigned to the Avaya SBCE, click **View** on the previous screen. The **System Information** window is displayed as shown below, containing the current device and the network settings. Note that the **A1** and **B1** interfaces correspond to the inside and outside interfaces for the Avaya SBCE, as shown in **Figure 1** in **Section 3**. Note that in the shared lab environment, multiple IP addresses are assigned to these interfaces, but only the highlighted ones are relevant to these Application Notes.

System Information: Avaya_SBCE

General Configuration

Appliance Name Avaya_SBCE
Box Type SIP
Deployment Mode Proxy

Device Configuration

HA Mode No
Two Bypass Mode No

Network Configuration

IP	Public IP	Netmask	Gateway	Interface
10.5.5.72	10.5.5.72	255.255.255.0	10.5.5.254	A1
172.16.157.148	172.16.157.148	255.255.255.0	172.16.157.129	B1
10.5.5.73	10.5.5.73	255.255.255.0	10.5.5.254	A1
172.16.157.146	172.16.157.146	255.255.255.0	172.16.157.129	B1
172.16.157.145	172.16.157.145	255.255.255.0	172.16.157.129	B1

DNS Configuration

Primary DNS 192.168.10.100
Secondary DNS
DNS Location DMZ
DNS Client IP 10.5.5.72

Management IP(s)

IP 192.168.10.70

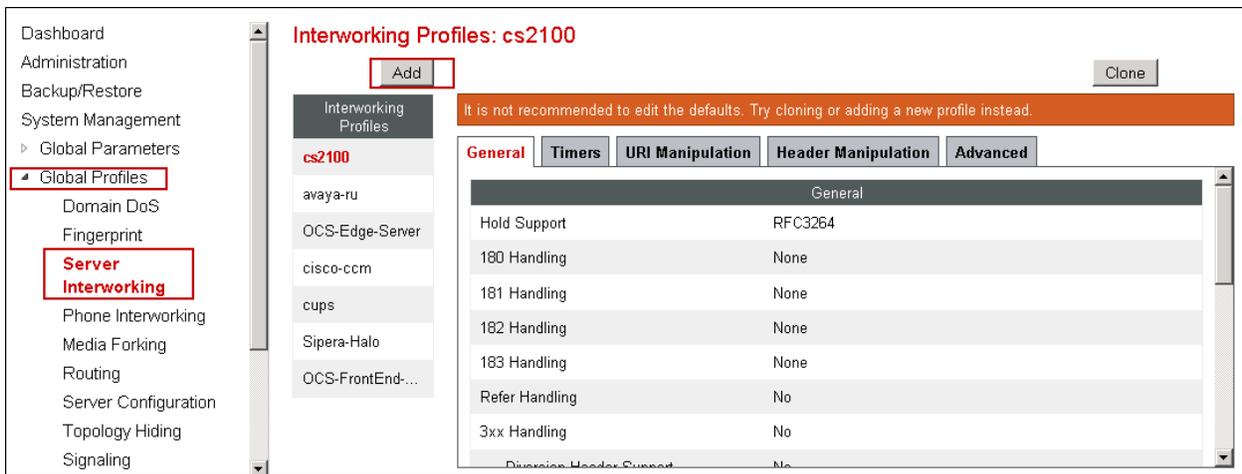
7.3. Global Profiles

The Global Profiles Menu, on the left navigation pane, allows the configuration of parameters across all Avaya SBCE appliances.

7.3.1. Server Interworking

Interworking Profile features are configured to facilitate the interoperability between the enterprise SIP-enabled solution (Call Server) and the SIP trunk service provider (Trunk Server). In the reference configuration, Session Manager functions as the Call Server and the Alestra SIP Proxy as the Trunk Server.

To configure the interworking profile in the enterprise direction, select **Global Profiles** → **Server Interworking** on the left navigation pane. Click **Add**.



Enter a descriptive name for the new profile. Click **Next**.



On the **General** screen, set Hold Support to RFC2543 and check the **T.38 Support** box. All other parameters retain their default values. Click **Next**.

The screenshot shows the 'Interworking Profile' configuration window, specifically the 'General' tab. The 'Hold Support' option is set to 'RFC2543 - c=0.0.0.0'. The 'T.38 Support' checkbox is checked. Other options include '180 Handling', '181 Handling', '182 Handling', '183 Handling', 'Refer Handling', '3xx Handling', 'Diversion Header Support', 'Delayed SDP Handling', 'URI Scheme', and 'Via Header Format'. The 'Back' and 'Next' buttons are visible at the bottom.

Parameter	Value
Hold Support	<input type="radio"/> None <input checked="" type="radio"/> RFC2543 - c=0.0.0.0 <input type="radio"/> RFC3264 - a=sendonly
180 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
181 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
182 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
183 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
Refer Handling	<input type="checkbox"/>
3xx Handling	<input type="checkbox"/>
Diversion Header Support	<input type="checkbox"/>
Delayed SDP Handling	<input type="checkbox"/>
T.38 Support	<input checked="" type="checkbox"/>
URI Scheme	<input checked="" type="radio"/> SIP <input type="radio"/> TEL <input type="radio"/> ANY
Via Header Format	<input checked="" type="radio"/> RFC3261 <input type="radio"/> RFC2543

Click **Next** on the **Privacy/DTMF** and **SIP Timers/Transport Timers** tabs (not shown). On the **Advanced Settings** tab, uncheck the **Topology Hiding: Change Call-ID** box and check the **AVAYA Extensions** box. Click **Finish** to save and exit.

Setting	Value
Record Routes	<input type="radio"/> None <input type="radio"/> Single Side <input checked="" type="radio"/> Both Sides
Topology Hiding: Change Call-ID	<input type="checkbox"/>
Call-Info NAT	<input type="checkbox"/>
Change Max Forwards	<input checked="" type="checkbox"/>
Include End Point IP for Context Lookup	<input type="checkbox"/>
OCS Extensions	<input type="checkbox"/>
AVAYA Extensions	<input checked="" type="checkbox"/>
NORTEL Extensions	<input type="checkbox"/>
Diversion Manipulation	<input type="checkbox"/>
Diversion Header URI	<input type="text"/>
Metaswitch Extensions	<input type="checkbox"/>
Reset on Talk Spurt	<input type="checkbox"/>
Reset SRTP Context on Session Refresh	<input type="checkbox"/>
Has Remote SBC	<input checked="" type="checkbox"/>
Route Response on Via Port	<input type="checkbox"/>
Cisco Extensions	<input type="checkbox"/>

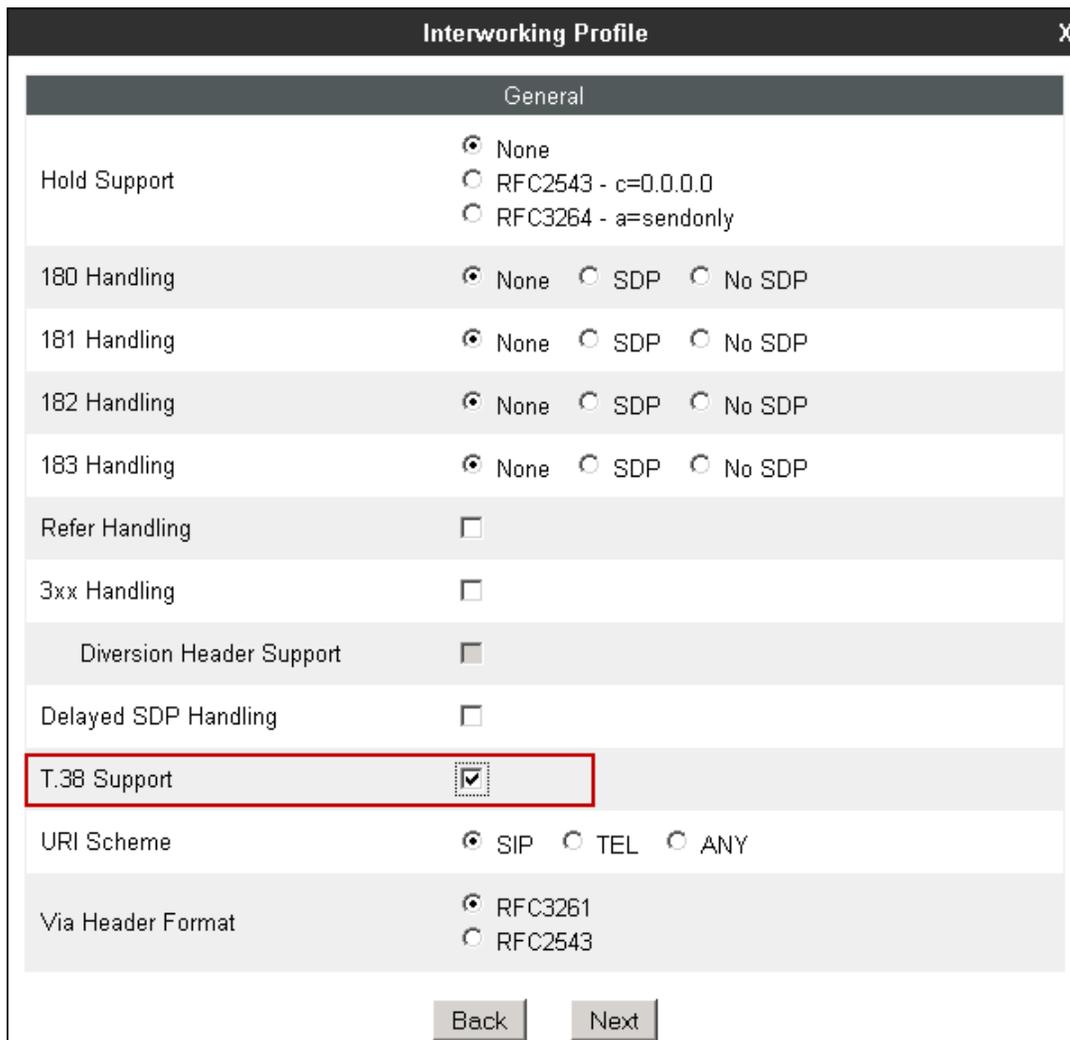
Back Finish

A second interworking profile named **Service Provider** in the direction of the SIP trunk to Alestra was similarly created. For this profile default values were used for all parameters except for **T.38 Support**, which was enabled.



The screenshot shows a dialog box titled "Interworking Profile" with a close button (X) in the top right corner. Inside the dialog, there is a text input field labeled "Profile Name" containing the text "Service Provider". Below the input field is a "Next" button.

General tab:



The screenshot shows the "Interworking Profile" dialog box with the "General" tab selected. The "T.38 Support" checkbox is checked and highlighted with a red box. Other options include "Hold Support", "180 Handling", "181 Handling", "182 Handling", "183 Handling", "Refer Handling", "3xx Handling", "Diversion Header Support", "Delayed SDP Handling", "URI Scheme", and "Via Header Format".

Option	Value
Hold Support	<input checked="" type="radio"/> None <input type="radio"/> RFC2543 - c=0.0.0.0 <input type="radio"/> RFC3264 - a=sendonly
180 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
181 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
182 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
183 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
Refer Handling	<input type="checkbox"/>
3xx Handling	<input type="checkbox"/>
Diversion Header Support	<input type="checkbox"/>
Delayed SDP Handling	<input type="checkbox"/>
T.38 Support	<input checked="" type="checkbox"/>
URI Scheme	<input checked="" type="radio"/> SIP <input type="radio"/> TEL <input type="radio"/> ANY
Via Header Format	<input checked="" type="radio"/> RFC3261 <input type="radio"/> RFC2543

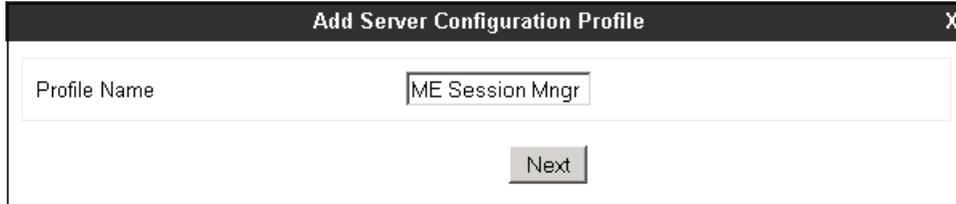
Advanced Settings tab:

Interworking Profile [X]

Record Routes	<input type="radio"/> None <input type="radio"/> Single Side <input checked="" type="radio"/> Both Sides
Topology Hiding: Change Call-ID	<input checked="" type="checkbox"/>
Call-Info NAT	<input type="checkbox"/>
Change Max Forwards	<input checked="" type="checkbox"/>
Include End Point IP for Context Lookup	<input type="checkbox"/>
OCS Extensions	<input type="checkbox"/>
AVAYA Extensions	<input type="checkbox"/>
NORTEL Extensions	<input type="checkbox"/>
Diversion Manipulation	<input type="checkbox"/>
Diversion Header URI	<input type="text"/>
Metaswitch Extensions	<input type="checkbox"/>
Reset on Talk Spurt	<input type="checkbox"/>
Reset SRTP Context on Session Refresh	<input type="checkbox"/>
Has Remote SBC	<input checked="" type="checkbox"/>
Route Response on Via Port	<input type="checkbox"/>
Cisco Extensions	<input type="checkbox"/>

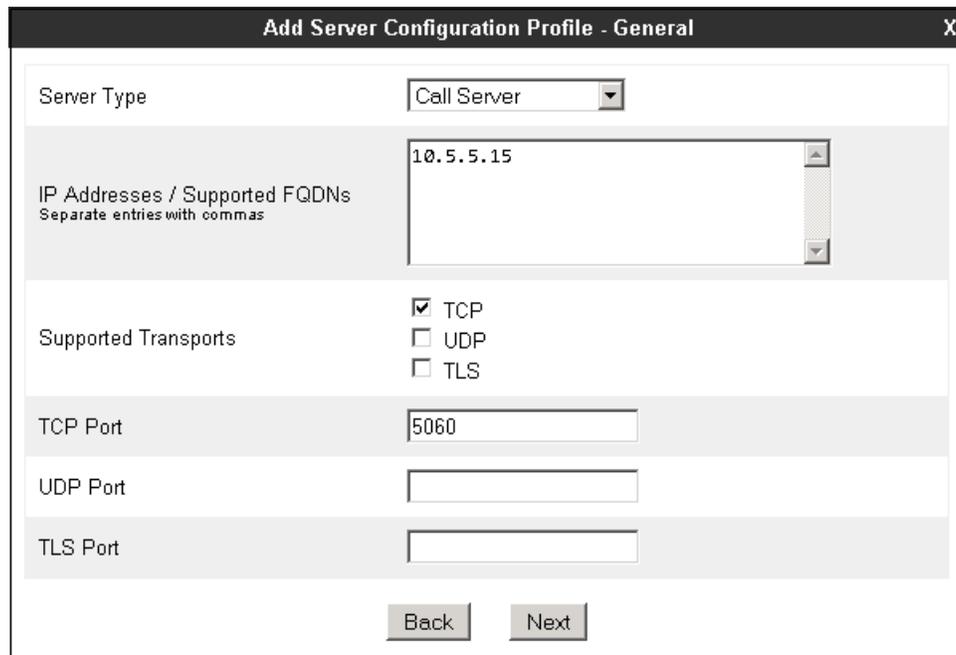
7.3.2. Server Configuration

Server Profiles are created to define the parameters for the Avaya SBCE two peers, i.e., Session Manager (Call Server) and the SIP Proxy at the service provider's network (Trunk Server). From the **Global Profiles** menu on the left-hand navigation pane, select **Server Configuration** and click the **Add** button (not shown) to add a new profile for the Call Server. Enter an appropriate **Profile Name** similar to the screen below. Click **Next**.



The screenshot shows a dialog box titled "Add Server Configuration Profile" with a close button (X) in the top right corner. Inside the dialog, there is a text input field labeled "Profile Name" containing the text "ME Session Mngr". Below the input field is a "Next" button.

On the **Add Server Configuration Profile - General** Tab select **Call Server** from the drop down menu for the **Server Type**. On the **IP Addresses / Supported FQDNs** field, enter the IP address of the Session Manager Security Module. Select **TCP** for **Supported Transports**, and enter **5060** under **TCP Port**. The transport protocol and port selected here must match the values defined for the Session Manager SIP entity in **Section 6.4**. Click **Next**.

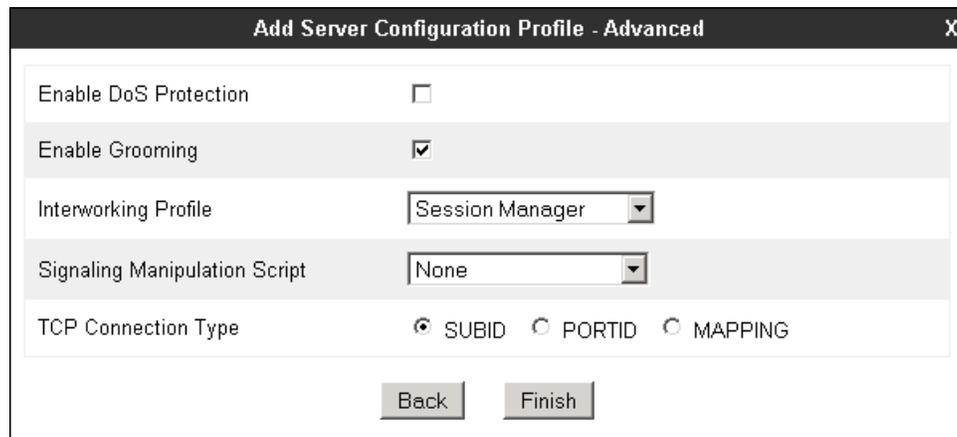


The screenshot shows a dialog box titled "Add Server Configuration Profile - General" with a close button (X) in the top right corner. The dialog contains several fields and options:

- Server Type:** A dropdown menu set to "Call Server".
- IP Addresses / Supported FQDNs:** A text area containing "10.5.5.15". Below the text area is the instruction "Separate entries with commas".
- Supported Transports:** Three radio buttons: "TCP" (checked), "UDP", and "TLS".
- TCP Port:** A text input field containing "5060".
- UDP Port:** An empty text input field.
- TLS Port:** An empty text input field.

At the bottom of the dialog are "Back" and "Next" buttons.

Click **Next** on the **Authentication** and **Heartbeat** tabs (not shown). On the **Advanced** tab, since TCP is used, check the **Enable Grooming** box. Select **Session Manager** from the **Interworking Profile** drop down menu. Click **Finish**.



The screenshot shows a dialog box titled "Add Server Configuration Profile - Advanced" with a close button (X) in the top right corner. The dialog contains the following settings:

- Enable DoS Protection:
- Enable Grooming:
- Interworking Profile:
- Signaling Manipulation Script:
- TCP Connection Type: SUBID PORTID MAPPING

At the bottom of the dialog are two buttons: "Back" and "Finish".

Similarly, to add the profile for the Trunk Server, click the **Add** button on the **Server Configuration** screen (not shown). Enter an appropriate **Profile Name** similar to the screen below. Click **Next**.



The screenshot shows a dialog box titled "Add Server Configuration Profile" with a close button (X) in the top right corner. The dialog contains the following settings:

- Profile Name:

At the bottom of the dialog is a "Next" button.

On the **Add Server Configuration Profile-General** Tab select *Trunk Server* from the drop down menu for the **Server Type**. On the **IP Addresses / Supported FQDNs** field, enter *192.168.145.9*, the IP Address of Alestra’s SIP proxy server. Select **UDP** for **Supported Transports**, and enter *5060* under **UDP Port**, as specified by Alestra.

Add Server Configuration Profile - General

Server Type: Trunk Server

IP Addresses / Supported FQDNs: 192.168.145.9
Separate entries with commas

Supported Transports: TCP, UDP, TLS

TCP Port: [Empty]

UDP Port: 5060

TLS Port: [Empty]

Back Next

Click **Next** on the **Authentication** and **Heartbeat** tabs (not shown). On the **Advanced** tab, select *Service Provider* from the **Interworking Profile** drop down menu. Click **Finish**

Add Server Configuration Profile - Advanced

Enable DoS Protection:

Enable Grooming:

Interworking Profile: Service Provider

Signaling Manipulation Script: None

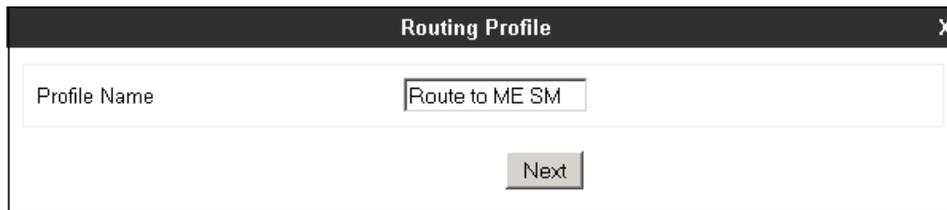
UDP Connection Type: SUBID, PORTID, MAPPING

Back Finish

7.3.3. Routing Profiles

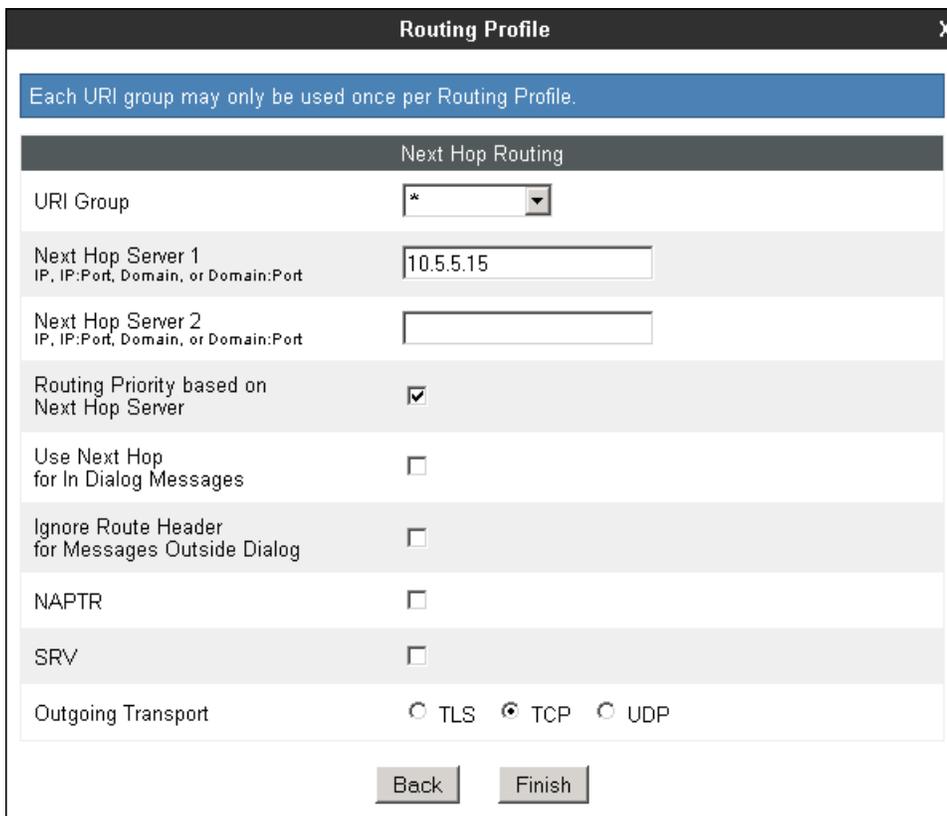
Routing profiles define a specific set of routing criteria that is used, in addition to other types of domain policies, to determine the path that the SIP traffic will follow as it flows through the Avaya SBCE interfaces.

Two Routing Profiles were created in the test configuration, one for inbound calls, with Session Manager as the destination, and the second one for outbound calls, which are routed to the Alestra SIP trunk. To create the inbound route, select the **Routing** tab from the **Global Profiles** menu on the left-hand side and select **Add** (not shown). Enter an appropriate **Profile Name** similar to the example below. Click **Next**.



The screenshot shows a window titled "Routing Profile" with a close button (X) in the top right corner. Below the title bar is a text input field labeled "Profile Name" containing the text "Route to ME SM". Below the input field is a "Next" button.

On the **Next Hop Routing** tab, enter the IP Address of Session Manager as **Next Hop Server 1**. Since the default well-known port value of 5060 for TCP was used, it is not necessary to enter the port number here. Check **Routing Priority based on Next Hop Server**. Choose **TCP** for **Outgoing Transport**. Click **Finish**.



The screenshot shows a window titled "Routing Profile" with a close button (X) in the top right corner. Below the title bar is a blue banner with the text "Each URI group may only be used once per Routing Profile." Below the banner is a tab labeled "Next Hop Routing". The form contains the following fields and options:

- URI Group: * (dropdown menu)
- Next Hop Server 1: 10.5.5.15 (text input field, with subtext "IP, IP:Port, Domain, or Domain:Port")
- Next Hop Server 2: (empty text input field, with subtext "IP, IP:Port, Domain, or Domain:Port")
- Routing Priority based on Next Hop Server:
- Use Next Hop for In Dialog Messages:
- Ignore Route Header for Messages Outside Dialog:
- NAPTR:
- SRV:
- Outgoing Transport: TLS TCP UDP

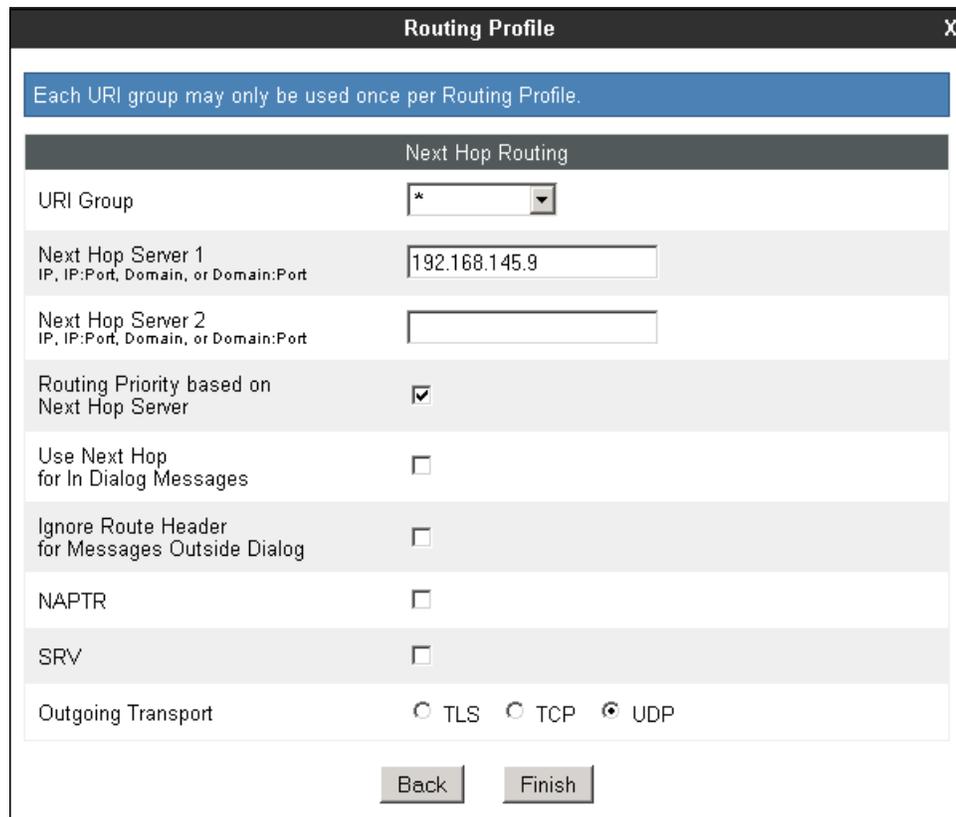
At the bottom of the form are "Back" and "Finish" buttons.

Back at the **Routing** tab, select **Add** (not shown) to repeat the process in order to create the outbound route. Enter an appropriate **Profile Name** similar to the example below. Click **Next**.



The screenshot shows a dialog box titled "Routing Profile" with a close button (X) in the top right corner. Inside the dialog, there is a text input field labeled "Profile Name" containing the text "Route to SP". Below the input field is a "Next" button.

On the Next Hop Routing tab, enter the IP Address of the service provider SIP proxy server as **Next Hop Server 1**. Since the default well-known port value of 5060 for UDP was used, it is not necessary to enter the port number here. Check **Routing Priority based on Next Hop Server**. Choose **UDP** for **Outgoing Transport**. Click **Finish**.



The screenshot shows the "Routing Profile" dialog box with the "Next Hop Routing" tab selected. A blue banner at the top states "Each URI group may only be used once per Routing Profile." The configuration options are as follows:

- URI Group: * (dropdown menu)
- Next Hop Server 1 (IP, IP:Port, Domain, or Domain:Port): 192.168.145.9
- Next Hop Server 2 (IP, IP:Port, Domain, or Domain:Port): (empty field)
- Routing Priority based on Next Hop Server:
- Use Next Hop for In Dialog Messages:
- Ignore Route Header for Messages Outside Dialog:
- NAPTR:
- SRV:
- Outgoing Transport: TLS TCP UDP

At the bottom of the dialog are "Back" and "Finish" buttons.

7.3.4. Topology Hiding

Topology Hiding is a security feature that allows the modification of several SIP headers, preventing private enterprise network information from being propagated to the untrusted public network.

Topology Hiding can also be used as an interoperability tool to adapt the host portion in the SIP headers to the IP addresses or domains expected on the service provider and the enterprise networks. For the compliance test, only the minimum configuration required to achieve interoperability on the SIP trunk was performed. Additional steps can be taken in this section to further mask the information that is sent from the enterprise to the public network.

To add the **Topology Hiding Profile** in the enterprise direction, select **Topology Hiding** from the **Global Profiles** menu on the left-hand side and click the **Add** button (not shown). Enter a **Profile Name** such as the one shown below. Click **Next**.



The screenshot shows a window titled "Topology Hiding Profile" with a close button (X) in the top right corner. Inside the window, there is a text input field labeled "Profile Name" containing the text "ME Sess Mngr". Below the input field is a "Next" button.

On the **Topology Hiding Profile** screen, click the **Add Header** button repeatedly to show the rest of the headers in the profile.



The screenshot shows the "Topology Hiding Profile" window with an "Add Header" button in the top right. Below it is a table with the following structure:

Header	Criteria	Replace Action	Overwrite Value
Request-Line	IP/Domain	Auto	

Below the table are "Back" and "Finish" buttons. A "Delete" link is also visible next to the table row.

For the **Request-Line**, **From** and **To** headers, select *Overwrite* in the **Replace Action** column and enter the enterprise SIP domain known by the Session Manager, *avayalab2.com*, in the **Overwrite Value** column of these headers, as shown below. Default values were used for all other fields. Click **Finish**.

Header	Criteria	Replace Action	Overwrite Value
Request-Line	IP/Domain	Overwrite	avayalab2.com
From	IP/Domain	Overwrite	avayalab2.com
To	IP/Domain	Overwrite	avayalab2.com
Record-Route	IP/Domain	Auto	
Via	IP/Domain	Auto	
SDP	IP/Domain	Auto	

A Topology Hiding profile named **Service Provider** was similarly configured in the direction of the SIP trunk to Alestra. During the compliance test, IP addresses instead of domains were used in all SIP messages between the Alestra SIP proxy server and the Avaya SBCE. Note that since the default action of *Auto* implies the insertion of IP addresses in the host portion of these headers, it was not necessary to modify any of the headers sent to the service provider.

Topology Hiding Profiles: Service Provider

Buttons: Add, Rename, Clone, Delete

Click here to add a description.

Header	Criteria	Replace Action	Overwrite Value
Request-Line	IP/Domain	Auto	---
Via	IP/Domain	Auto	---
SDP	IP/Domain	Auto	---
Record-Route	IP/Domain	Auto	---
To	IP/Domain	Auto	---
From	IP/Domain	Auto	---

Buttons: Edit

7.4. Domain Policies

Domain Policies allow the configuration of sets of rules designed to control and normalize the behavior of call flows, based upon various criteria of communication sessions originating from or terminating in the enterprise. Domain Policies include rules for Application, Media, Signaling, Security, etc.

In the reference configuration, only one new Signaling Rule was defined. All other rules under Domain Policies, linked together on End Point Policy Groups later in this section, used one of the default sets already pre-defined in the configuration. Please note that changes should not be made to any of the defaults. If changes are needed, it is recommended to create a new rule by cloning one of the defaults and then make the necessary changes to the new rule.

7.4.1. Signaling Rules

A Signaling Rule was created in the sample configuration to remove (block) the following headers:

- AV-Global-Session-ID
- Alert-Info
- P-AV-Message-ID
- P-Location
- P-Charging-Vector

These headers are sent in SIP messages from the Session Manager to the Avaya SBCE. They contain private IP addresses and SIP Domains from the enterprise, which should not be propagated outside of the enterprise boundaries.

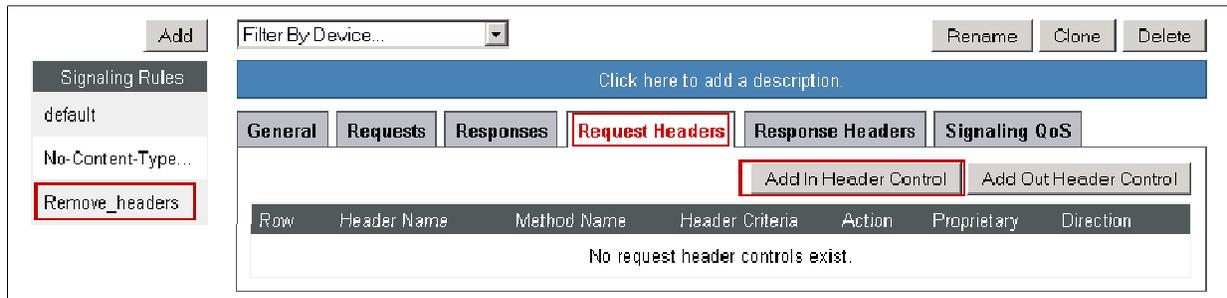
In the **Domain Policies** menu on the left-hand side, select **Signaling Rules**, then **Add Rule** (not shown). Enter an appropriate name like in the example below. Click **Next**.



The screenshot shows a dialog box titled "Signaling Rule" with a close button (X) in the top right corner. Inside the dialog, there is a text input field labeled "Rule Name" containing the text "Remove_headers". Below the input field is a "Next" button.

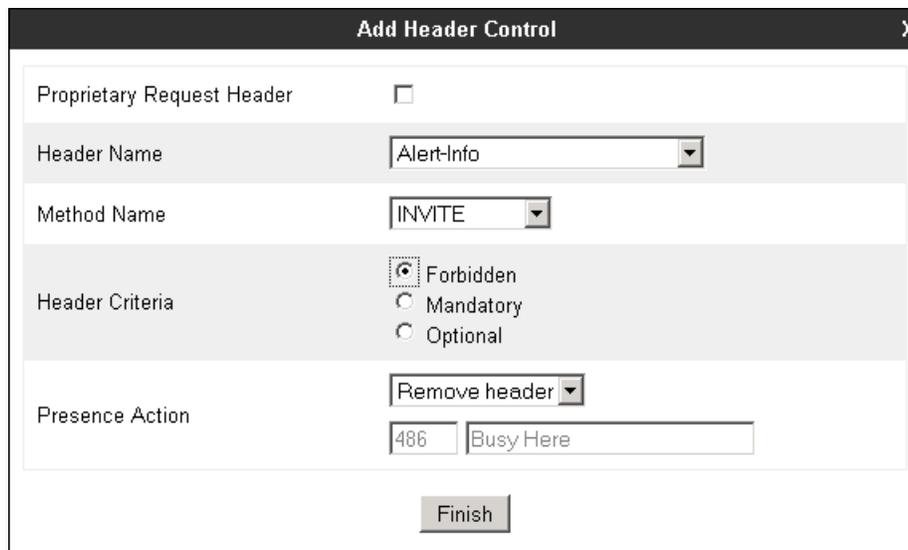
On the next three pages (not shown), leave sections **Inbound**, **Outbound** and **Content-Type Policies** with their default values. Click **Next**. On the **Signaling QoS** tab, default values were used. Click **Finish**.

On the newly created **Remove_headers** Signaling Rule, select the **Request Headers** tab to create the manipulations performed on request messages. Select **Add In Header Control**.



In the **Add Header Control** screen select the following:

- **Header Name: Alert-Info**
- **Method Name: INVITE**
- **Header Criteria: Forbidden**
- **Presence Action: Remove Header**
- **Click Finish**



Select **Add In Header Control** as needed to configure the remaining header control rules for the AV-Global-Session-ID, P-AV-Message-ID, P-Location and P-Charging-Vector headers. For these headers, make sure to check the **Proprietary Request Header** box in the **Add Header Control** tab. This will allow typing the name of the specific header on the **Header Name** box. Once completed, the **Request Headers** tab should look like the following screen.

Row	Header Name	Method Name	Header Criteria	Action	Proprietary	Direction		
1	AV-Global-Session-ID	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
2	Alert-Info	ALL	Forbidden	Remove Header	No	IN	Edit	Delete
3	P-AV-Message-ID	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
4	P-Charging-Vector	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
5	P-Location	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete

Select the **Response Headers** tab to similarly create the manipulations performed on response messages. Select **Add In Header Control** (not shown).

The screen below shows the settings for the Alert-Info header on response messages.

Add Header Control X

Proprietary Response Header

Header Name

Response Code

Method Name

Header Criteria
 Forbidden
 Mandatory
 Optional

Presence Action

Select **Add In Header Control** as needed to configure the remaining header control rules for the AV-Global-Session-ID, P-AV-Message-ID, P-Location and P-Charging-Vector headers. For these headers, make sure to check the **Proprietary Request Header** box in the **Add Header Control** tab. This will allow typing the name of the specific header on the **Header Name** box. Once completed, the **Response Headers** tab should look like the following screen.

General		Requests	Responses	Request Headers	Response Headers	Signaling QoS			
					Add In Header Control	Add Out Header Control			
Row	Header Name	Response Code	Method Name	Header Criteria	Action	Proprietary	Direction		
1	AV-Global-Session-ID	200	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
2	Alert-Info	200	ALL	Forbidden	Remove Header	No	IN	Edit	Delete
3	P-AV-Message-ID	200	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
4	P-Charging-Vector	200	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
5	P-Location	200	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete

7.4.2. End Point Policy Groups

End Point Policy Groups associate the different sets of rules (Media, Signaling, Security, etc) to be applied to specific SIP messages traversing through the Avaya SBCE.

To create an End Point Policy Group for the enterprise, select **End Point Policy Groups** under the **Domain Policies** menu. Select **Add** (not shown).

Enter an appropriate name in the **Group Name** field. *Enterprise* was used. Click **Next**.

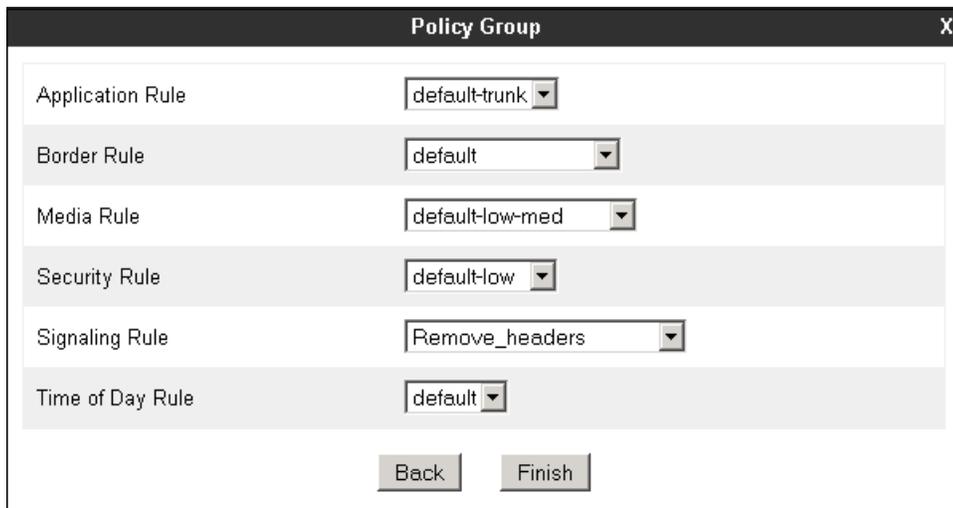


Policy Group

Group Name: Enterprise

Next

In the Policy Group tab, all fields used one of the default sets already pre-defined in the configuration, with the exception of the **Signaling Rule**, where the *Remove_headers* rule created in **Section 7.4.1** was selected. Click **Finish**.



Policy Group

Application Rule: default-trunk

Border Rule: default

Media Rule: default-low-med

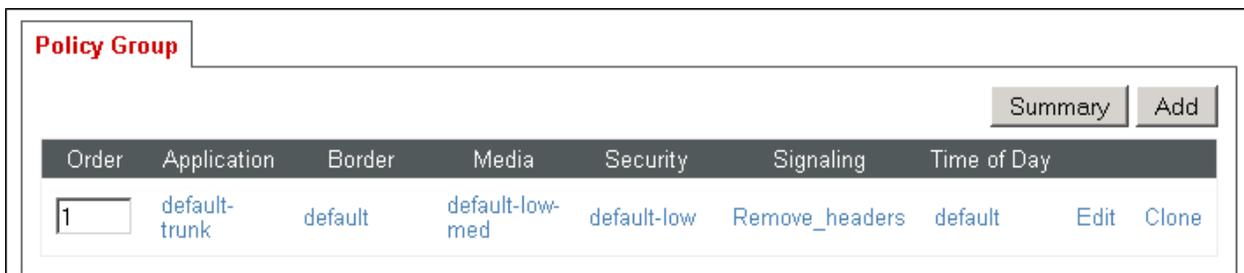
Security Rule: default-low

Signaling Rule: Remove_headers

Time of Day Rule: default

Back Finish

The screen below shows the **Enterprise** End Point Policy Group after the configuration was completed.



Policy Group

Summary Add

Order	Application	Border	Media	Security	Signaling	Time of Day		
1	default-trunk	default	default-low-med	default-low	Remove_headers	default	Edit	Clone

A second End Point Policy Group was created for the service provider, repeating the steps described above. Defaults were used in this case for all fields. The screen below shows the **Service Provider** End Point Policy Group after the configuration was completed.

Policy Group								Summary	Add
Order	Application	Border	Media	Security	Signaling	Time of Day			
1	default-trunk	default	default-low-med	default-low	default	default	Edit	Clone	

7.5. Device Specific Settings

The **Device Specific Settings** determine server specific parameters that determine how the device will work when deployed on the network. Among the parameters defined here are IP addresses, media and signaling interfaces, call flows, etc.

7.5.1. Network Management

The network configuration parameters should have been previously specified during installation of the Avaya SBCE. In the event that changes need to be made to the network configuration, they can be made here.

Select **Network Management** from **Device Specific Settings** on the left-side menu (not shown). Under **Devices** in the centre pane, select the device being managed, **Avaya_SBCE** in the sample configuration. On the **Network Configuration** tab, verify or enter the network information as needed. Note that the **A1** interface is used for the internal side and **B1** is used for the external side of the Avaya SBCE.

Network Management: Avaya_SBCE

Devices

Avaya_SBCE

Network Configuration

Interface Configuration

Modifications or deletions of an IP address or its associated data require an application restart before taking effect. Application restarts can be issued from [System Management](#).

A1 Netmask	A2 Netmask	B1 Netmask	B2 Netmask
<input type="text" value="255.255.255.0"/>	<input type="text"/>	<input type="text" value="255.255.255.0"/>	<input type="text"/>
<input type="button" value="Add"/>		<input type="button" value="Save"/> <input type="button" value="Clear"/>	

IP Address	Public IP	Gateway	Interface
<input type="text" value="10.5.5.72"/>	<input type="text"/>	<input type="text" value="10.5.5.254"/>	<input type="text" value="A1"/> <input type="button" value="Delete"/>
<input type="text" value="172.16.157.148"/>	<input type="text"/>	<input type="text" value="172.16.157.129"/>	<input type="text" value="B1"/> <input type="button" value="Delete"/>

On the **Interface Configuration** tab, verify the **Administrative Status** is **Enabled** for both the **A1** and **B1** interfaces. Click the **Toggle** buttons if necessary to enable the interfaces.

Network Configuration		Interface Configuration	
Name	Administrative Status		
A1	Enabled		Toggle
A2	Disabled		Toggle
B1	Enabled		Toggle
B2	Disabled		Toggle

7.5.2. Media Interface

Media Interfaces were created to specify the IP address and port range in which the Avaya SBCE will accept media streams on each interface. Packets leaving the interfaces of the Avaya SBCE will advertise this IP address and one of the ports in this range as the listening IP address and port in which it will accept media from the Call or Trunk Server.

To add the Media Interface in the enterprise direction, select **Media Interface** from the **Device Specific Settings** menu on the left-hand side, select the **Avaya_SBCE** device and click the **Add** button (not shown). On the **Add Media Interface** screen, enter an appropriate **Name** for the Media Interface. Select the private IP Address for the Avaya SBCE from the **IP Address** drop-down menu. The **Port Range** was left at the default values of **35000-40000**. Click **Finish**.

Add Media Interface X

Name

IP Address

Port Range -

A second Media Interface facing the public network side was similarly created with the name **Public_med**, as shown below. The outside IP Address of the Avaya SBCE was selected from the drop-down menu. The **Port Range** was left at the default values.

Add Media Interface X

Name

IP Address

Port Range -

Once the configuration is complete, the **Media Interface** screen will appear as follows.

Media Interface: Avaya_SBCE

Devices

Avaya_SBCE

Media Interface

Modifying or deleting an existing media interface will require an application restart before taking effect. Application restarts can be issued from [System Management](#).

Name	Media IP	Port Range	
Private_med	10.5.5.72	35000 - 40000	Edit Delete
Public_med	172.16.157.148	35000 - 40000	Edit Delete

7.5.3. Signaling Interface

Signaling Interfaces are created to specify the IP addresses and ports in which the Avaya SBCE will listen for signaling traffic in both the inside and outside networks.

To add the Signaling Interface in the enterprise direction, select **Signaling Interface** from the **Device Specific Settings** menu on the left-hand side, select the **Avaya_SBCE** device and click the **Add** button (not shown). On the **Add Signaling Interface** screen, enter an appropriate **Name** for the interface. Select the private IP Address for the Avaya SBCE from the **IP Address** drop-down menu. Enter **5060** for **TCP Port**, since TCP port 5060 is used to listen to signaling traffic from Session Manager in the sample configuration. Click **Finish**.

Add Signaling Interface X

Name	<input type="text" value="Private_sig"/>
IP Address	<input type="text" value="10.5.5.72"/>
TCP Port <small>Leave blank to disable</small>	<input type="text" value="5060"/>
UDP Port <small>Leave blank to disable</small>	<input type="text"/>
Enable Stun	<input type="checkbox"/>
TLS Port <small>Leave blank to disable</small>	<input type="text"/>
TLS Profile	<input type="text" value="AvayaSBCServer"/>
Enable Shared Control	<input type="checkbox"/>
Shared Control Port	<input type="text"/>

A second Signaling Interface with the name **Public_sig** was similarly created in the network direction. The outside IP Address of the Avaya SBCE was selected from the drop-down menu. **UDP Port 5060** was selected since this is the protocol and port used by the Avaya SBCE to listen to the service provider's SIP traffic.

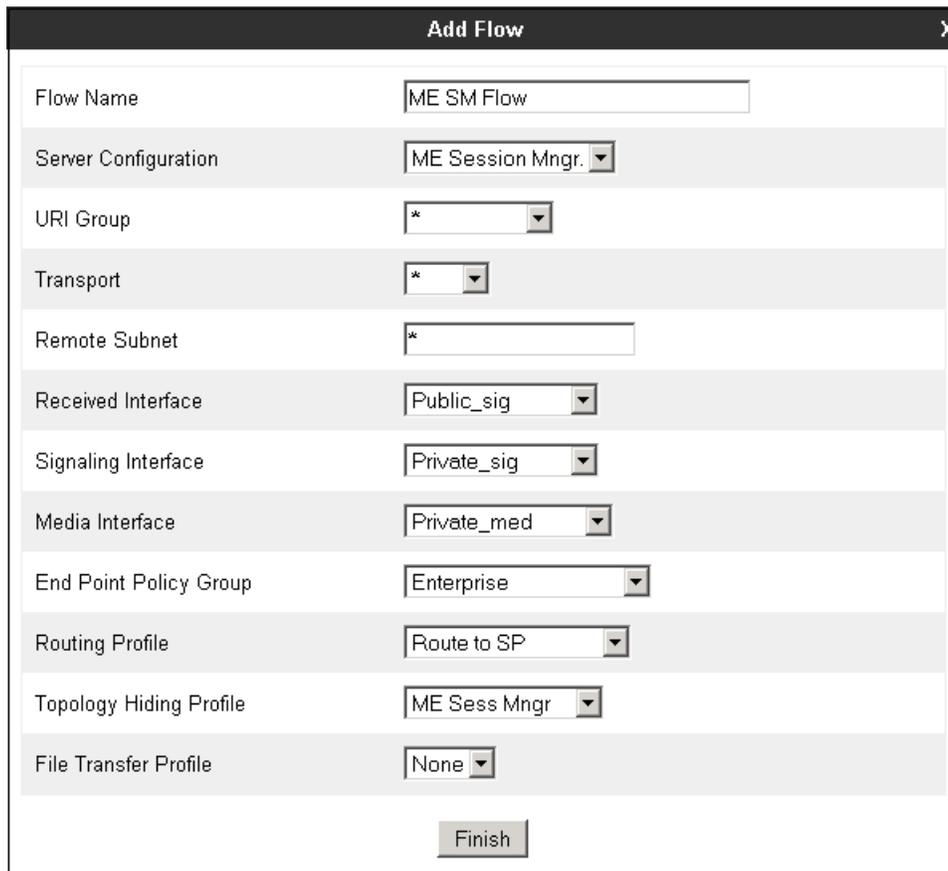
Once the configuration is complete, the **Signaling Interface** screen will appear as follows:

Name	Signaling IP	TCP Port	UDP Port	TLS Port	TLS Profile	
Private_sig	10.5.5.72	5060	---	---	None	Edit Delete
Public_sig	172.16.157.148	---	5060	---	None	Edit Delete

7.5.4. End Point Flows

End Point Flows determine the path to be followed by the packets traversing through the Avaya SBCE. They also combine the different sets of rules and profiles previously configured, to be applied to the SIP traffic traveling in each direction.

To create the call flow toward the enterprise, from the **Device Specific** menu, select **End Point Flows**, then select the **Server Flows** tab. Click **Add** (not shown). The screen below shows the flow named **ME SM Flow** created in the sample configuration. The flow uses the interfaces, policies, and profiles defined in previous sections. Note the **Routing Profile** selection, which is the reverse route of the flow. Click **Finish**.



The screenshot shows a configuration window titled "Add Flow" with a close button (X) in the top right corner. The window contains several fields and dropdown menus for configuring a flow. The fields are as follows:

Field Name	Value
Flow Name	ME SM Flow
Server Configuration	ME Session Mngr.
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Public_sig
Signaling Interface	Private_sig
Media Interface	Private_med
End Point Policy Group	Enterprise
Routing Profile	Route to SP
Topology Hiding Profile	ME Sess Mngr
File Transfer Profile	None

At the bottom center of the window is a button labeled "Finish".

A second Server Flow with the name **SIP Trunk Flow** was similarly created in the network direction. The flow uses the interfaces, policies, and profiles defined in previous sections. Note the **Routing Profile** selection, which is the reverse route of the flow. Click **Finish**.

Add Flow X

Flow Name

Server Configuration

URI Group

Transport

Remote Subnet

Received Interface

Signaling Interface

Media Interface

End Point Policy Group

Routing Profile

Topology Hiding Profile

File Transfer Profile

The two Server Flows created in the sample configuration are summarized on the screen below:

Devices	Subscriber Flows	Server Flows					
Avaya_SBCE	Click here to add a row description.						
Server Configuration: ME Session Mngr.							
Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile	
1	ME SM Flow	*	Public_sig	Private_sig	Enterprise	Route to SP	
View Clone Edit Delete							
Server Configuration: Service Provider							
Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile	
1	SIP Trunk Flow	*	Private_sig	Public_sig	Service Provider	Route to ME SM	
View Clone Edit Delete							

8. Alestra Enlace IP SIP Trunk Service Configuration

Alestra is responsible for the configuration of the Enlace IP SIP Trunk Service on its network. The customer will need to provide the IP address used to reach the Avaya SBCE at the enterprise. Alestra will provide the customer the necessary information to configure the SIP connection from the enterprise site to the Alestra network, including:

- IP address of the Alestra SIP Proxy server.
- Supported codecs and order of preference.
- DID numbers.
- All IP addresses and port numbers used for signaling or media that will need access to the enterprise network through any security devices.

This information is used to complete the configuration of Communication Manager, Session Manager and the Avaya SBCE discussed in the previous sections.

9. Verification and Troubleshooting

This section provides verification steps that may be performed in the field to verify that the solution is configured properly. This section also provides a list of useful troubleshooting commands that can be used to troubleshoot the solution.

9.1. General Verification Steps

- Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active for more than 35 seconds. This time period is included to verify that proper routing of the SIP messaging has satisfied SIP protocol timers.
- Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active for more than 35 seconds.
- Verify that the user on the PSTN can end an active call by hanging up.
- Verify that an endpoint at the enterprise site can end an active call by hanging up.

9.2. Communication Manager Verification

The following commands can be entered in the Communication Manager SAT terminal to verify the SIP trunk functionality:

- **list trace station** <extension number>
Traces calls to and from a specific station.
- **list trace tac** <trunk access code number>
Trace calls over a specific trunk group.
- **status signaling-group** <signaling group number>
Displays signaling group service state.
- **status trunk** <trunk group number>
Displays trunk group service state.
- **status station** <extension number>
Displays signaling and media information for an active call on a specific station.

9.3. Session Manager Verification

Log in to System Manager. Under the **Elements** section, navigate to **Session Manager** → **System Status** → **SIP Entity Monitoring**. Verify that the state of the Session Manager links under the **Conn. Status** and **Link Status** columns to Communication Manager and the Avaya SBCE is **UP**, like shown on the screen below.

Home / Elements / Session Manager / System Status / SIP Entity Monitoring Help ?

Session Manager Entity Link Connection Status

This page displays detailed connection status for all entity links from a Session Manager.

All Entity Links for Session Manager: [SessionManager1](#)

Status Details for the selected Session Manager:

5 Items | Refresh Filter: Enable

	SIP Entity Name	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
<input type="radio"/>	CommunicationManager1	10.5.5.12	5060	TCP	FALSE	UP	200 OK	UP
<input type="radio"/>	Presence1	10.5.5.16	5061	TLS	FALSE	UP	200 OK	UP
<input type="radio"/>	Messaging	10.5.5.12	6060	TCP	FALSE	UP	200 OK	UP
<input type="radio"/>	Avaya SBCE	10.5.5.72	5060	TCP	FALSE	UP	200 OK	UP
<input type="radio"/>	C.M. Trunk 2	10.5.5.12	5062	TCP	FALSE	UP	200 OK	UP

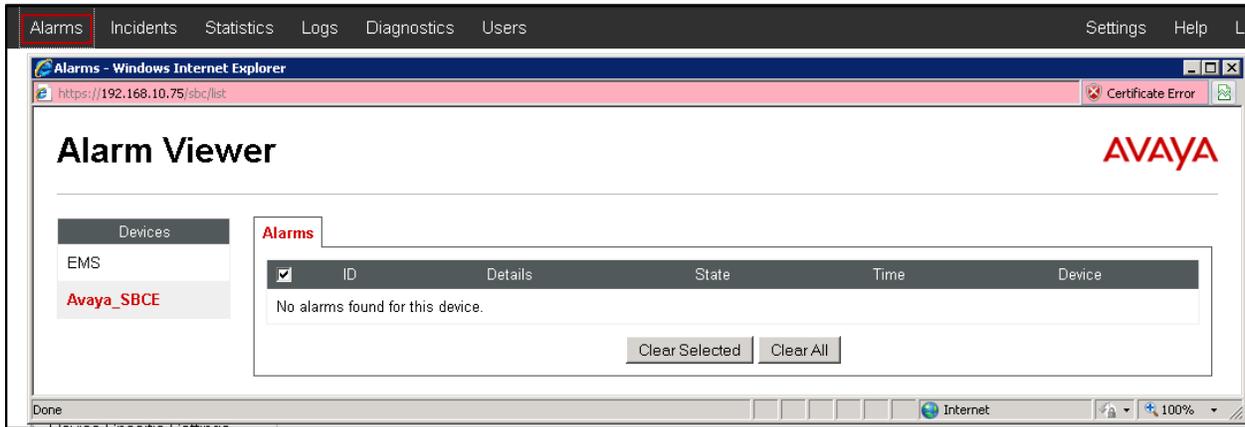
Other Session Manager useful verification and troubleshooting tools include:

- **traceSM** – Session Manager command line tool for traffic analysis. Login to the Session Manager command line management interface to run this command.
- **Call Routing Test** - The Call Routing Test verifies the routing for a particular source and destination. To run the routing test, from the System Manager Home screen navigate to **Elements** → **Session Manager** → **System Tools** → **Call Routing Test**. Enter the requested data to run the test

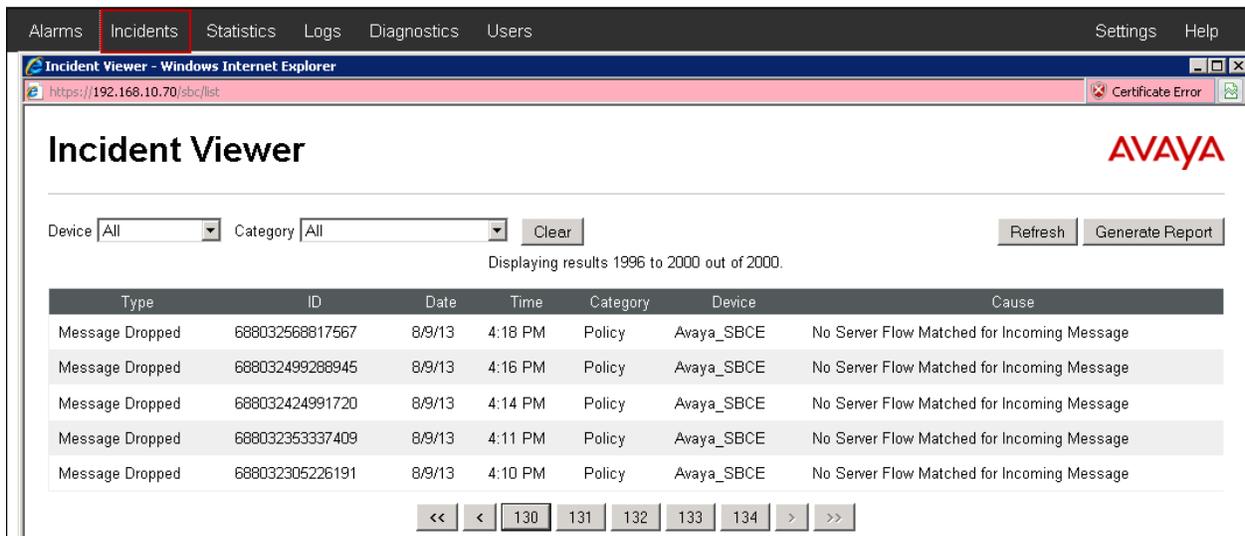
9.4. Avaya SBCE Verification

There are several links and menus located on the taskbar at the top of the screen of the web interface that can provide useful diagnostic or troubleshooting information.

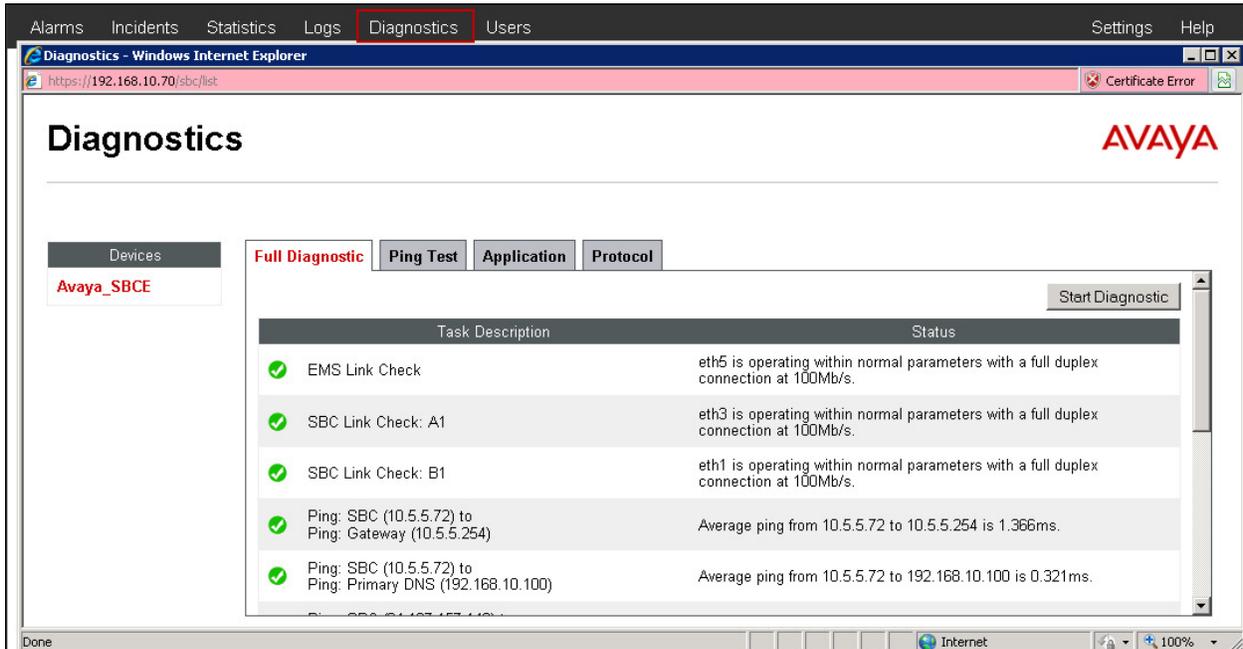
Alarms: Provides information about the health of the SBC.



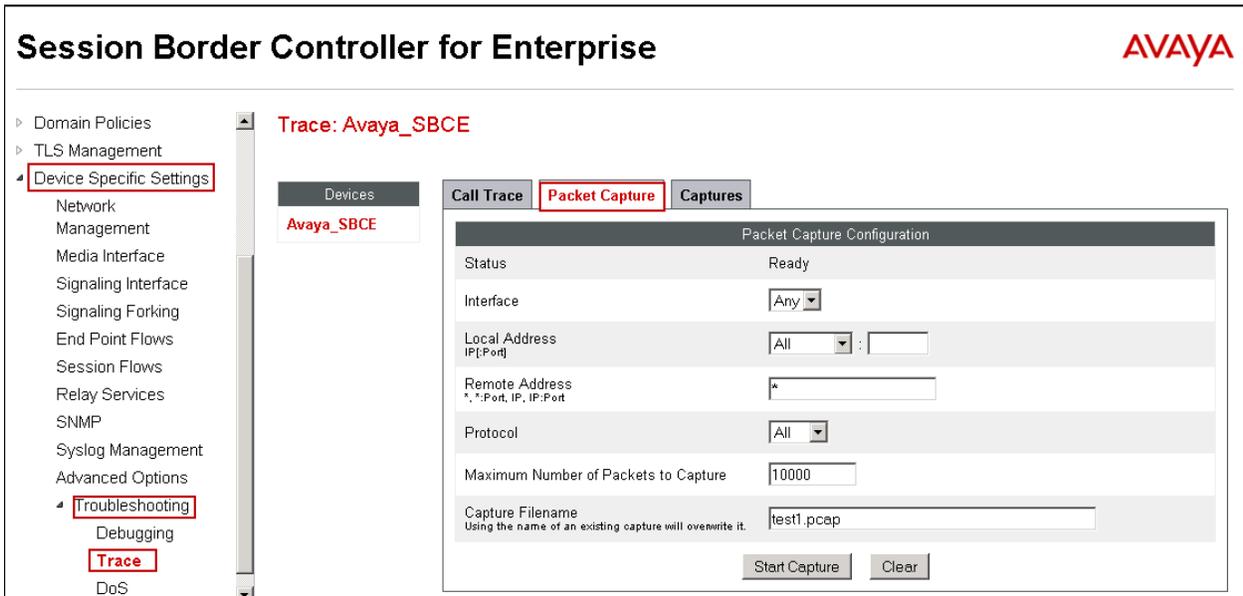
Incidents : Provides detailed reports of anomalies, errors, policies violations, etc.



Diagnostics: This screen provides a variety of tools to test and troubleshoot the SBC network connectivity.



Additionally, the Avaya SBCE contains an internal packet capture tool that allows the capture of packets on any of its interfaces, saving them as *pcap* files. Navigate to **Device Specific Settings** → **Troubleshooting** → **Trace**. Select the **Packet Capture** tab, set the desired configuration for the trace and click **Start Capture**.



Once the capture is stopped, click the **Captures** tab and select the proper *pcap* file. Note that the date and time is appended to the filename specified previously. The file can now be saved to the local PC, where it can be opened with an application such as Wireshark.

Call Trace	Packet Capture	Captures
Refresh		
File Name	File Size (bytes)	Last Modified
test1_20130830102339.pcap	393,216	August 30, 2013 10:24:04 AM GMT
		Delete

10. Conclusion

These Application Notes describe the procedures required to configure an Avaya Aura® Solution for Midsize Enterprise 6.2.2 and Avaya Session Border Controller for Enterprise 6.2, to connect to the Alestra Enlace IP SIP Trunk Service, as shown in **Figure 1**.

Interoperability testing of the sample configuration was completed with successful results for all test cases with the exception of the observations/limitations described in **Section 2.2**.

11. References

This section references the documentation relevant to these Application Notes. Additional Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Installing and Configuring Avaya Aura® System Platform*, Release 6.3, May 2013.
- [2] *Administering Avaya Aura® System Platform*, Release 6.3, May 2013.
- [3] *Avaya Aura® Solution for the Midsize Enterprise (ME) 6.2.2 Intelligent Workbook*, Version 1.3, August 2013
- [4] *Implementing Avaya Aura® Solution for the Midsize Enterprise*, Release 6.2.2, August 2013
- [5] *Upgrading Avaya Aura® Solution for Midsize Enterprise*, Release 6.2.2, August 2013
- [6] *Avaya Aura® Solution for the Midsize Enterprise 6.2.2 Release Notes*, August 2013
- [7] *Administering Avaya Aura® Communication Manager*, Release 6.3, May 2013, Document Number 03-300509.
- [8] *Avaya Aura® Communication Manager Feature Description and Implementation*, Release 6.3, May 2013, Document Number 555-245-205.
- [9] *Administering Avaya Aura® Session Manager*, Release 6.3, June 2013.
- [10] *Installing Avaya Session Border Controller for Enterprise*, Release 6.2, June 2013
- [11] *Administering Avaya Session Border Controller for Enterprise*, Release 6.2, March 2013
- [12] *Administering Avaya one-X® Communicator*, December 2012.
- [13] *Using Avaya one-X® Communicator, Release 6.1*, October 2011.
- [14] *Implementing Avaya Flare® Experience for Windows*. Release 1.1 February 2013.
- [15] *RFC 3261 SIP: Session Initiation Protocol*, <http://www.ietf.org/>.
- [16] *RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, <http://www.ietf.org/>

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